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IP Office Release 7.0

Product Description

15-601041 Issue 22.a.- (21 March 2011 - 11:21)

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Documentation information

For the most current versions of documentation, go to the Avaya Support web site (http://www.avaya.com/support) or the IP Office Knowledge Base (http://marketingtools.avaya.com/knowledgebase/).

Avaya Support

Avaya provides a telephone number for you to use to report problems or to ask questions about your contact center. The support telephone number is 1 800 242 2121 in the United States. For additional support telephone numbers, see the Avaya Web site: http://www.avaya.com/support.

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Chapter 1. Introduction

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1. Introduction

This document is not a contractual document. The specification of the Avaya IP Office family is subject to change without notice. Not all components and features documented are available in all territories; refer to your Avaya representative for further details. This document should be read in conjunction with any issued technical bulletins and/or product offer announcements.

Avaya IP Office

Simple, Powerful Communications for Small Businesses.

What is IP Office?

IP Office is a versatile communications solution that combines the reliability and ease of a traditional telephony system with the applications and advantages of an IP telephony solution. This converged communications solution can help businesses reduce costs, increase productivity, and improve customer service.

How can I use it in my business?

Avaya IP Office unifies your communications, providing your employees with a solution that lets them handle all their business communications on the device of their choice: their laptop, mobile phone, office telephone or home telephone; using wired, wireless or broadband connections.

What are my choices?

Successful companies are always looking for new ways to do more with less – keeping their business flexible, innovative and competitive without driving up operating costs and capital expenses. Avaya IP Office is the right choice for any small business today – whether you have 5, 25, 250 or 1000 employees… just getting started or already established… have a single location or multiple locations across the globe

With IP Office you can tailor the solution to your user needs (e.g. Teleworker, Mobile Worker, Customer Service Agent, etc.). Whether you have 5 employees, 25, 250 or more, IP Office is the right choice.

IP Office: Three key things to know

All small and medium size businesses are seeking ways to reduce costs and improve the way it operates. Like every business, you're looking to keep all your customers, add new ones and grow at the pace that's right for you. Avaya understands this. With over one hundred years of experience as a leader in communications, we know that the right solution for your business is one that helps you increase profitability, improve productivity and gain competitive advantages.

Serving Customers Effectively with Unified Communications

Your office telephone, cell phone, e-mail, texting, instant messaging and even your home telephone are all essential to the way you work today. Unlike other solutions, Avaya IP Office uses built-in intelligence to simplify your use of wired, wireless and Internet communications. Instead of keeping your communications separate, Avaya IP Office brings them together so you can easily turn a home or mobile phone into an office extension, collaborate with dozens of customers or staff members on a conference call and get detailed reports that show how well your company is responding to customers. With rich functionality, IP Office naturally creates new ways of thinking about the role communications can play in your business.

See what Avaya can do for you

You need a communications system – every business does. To find one that's right for your business, start with Avaya. With solutions like IP Office, we're revolutionizing how small and medium size businesses communicate. Now is the time to see what an Avaya solution can do for your business.

Reduce monthly costs. Now.

IP Office will help you lower the cost of communications, with capabilities like conferencing, making calls over a managed Internet service (Voice over IP) and the "all-in-one" benefits of a converged communications system.

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Leave the office. Be accessible.

With easy, flexible options for call/message forwarding and one-number reachability, IP Office keeps everyone in touch. Get the freedom to go where you want and never miss important business calls.

Serve better. Sell more.

IP Office can give you a customer sales and service center designed for your needs and your budget – with all the routing and reporting capabilities you need. Deliver the personal service that builds sales and loyalty.

Get connected.

Talk to your Avaya Business Partner. Discuss where you want communications to add value to your business. Learn about the different service and support options that are available. See why thousands of growing businesses rely on the innovative Avaya IP Office solution.

The right choice for you and your business.

How we communicate is a personal choice – it has to match the needs of your business. And your needs change depending on whether your employees are working in the office, at home, or on the road. That's why when you choose IP Office you can also choose from a whole range of communication tools and applications designed to boost productivity. Choose a basic telephone or one with all the "bells and whistles". Connect our IP telephones directly to your office LAN – you can even take them home and get all the features you have at the office. With Avaya IP Office, a PC or laptop and an internet connection you can turn the screen of your PC into a telephone. And our wireless solutions make it easier to roam the office. With all of our IP Office capabilities, our goal is to make your communications simple and cost-effective. Let your Avaya BusinessPartner put together a selection of tools and applications that's right for you.

Fine-tuning performance.

How many calls are you handling an hour, a day? What are your peak calling periods? How many calls typically turn into sales? The Avaya IP Office reporting capabilities can help you measure and manage your availability and response to customers.

Day-to-day administration.

Once your system is up and running you will benefit from the graphical administration tools that simplify day-to-day tasks, such as updating directories and moving telephone extensions.

Getting started.

Is your communications network ready for IP Office? We'll make sure. Avaya has created assessment and automated configuration tools to make sure that when your system is installed it's ready to meet your needs starting Day One.

Keeping ongoing management simple.

Concerned about needing extra resources to administer a system as powerful as IP Office? There's no need for worry. IP Office comes with a whole set of graphical tools to keep ongoing management simple.

Does my current phone system give my business what it needs?

If it is based on old technology, probably not. Your competitors will react faster and appear more professional with the latest in communications software. IP Office delivers the capabilities that allow you to keep up with or overtake the competition.

Do I need to understand the technology to implement it?

No. IP Office is designed specifically to give you more functionality without making more demands on your resources. Rely on your certified Avaya BusinessPartner for support before, during and after your purchase. We' ll take care of you so you don't have to worry.

Do I need to spend a lot?

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Not at all. You have choices based on your budget needs. Easy leasing or financing plans not only make this affordable; they help you quickly cut monthly expenses immediately. And you only have to buy/lease what you need, when you need it.

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Is IP technology established and reliable?

Yes. With over 170,000 systems and 6 Million users worldwide Avaya IP Office has the track record businesses like yours can rely on. Customers like you are saving money and boosting productivity and many are managing IP Office themselves.

I have old systems but am adding an office. Should I consider newer technology?

Not only would this be a way for you to experience the rich functionality of the latest communications applications, but we may be able to network with your existing equipment, as well as provide a gradual migration plan for your other locations.

I have a Nortel BCM or Nortel Norstar system. Can I keep my existing telephones?

Yes. IP Office supports the majority of Nortel BCM and Norstar telephones. You can save between 40 to 60% of your original investment by keeping your existing telephones. If you want to add additional telephones or replace older IP Office supports nearly every mix of telephones.

How quickly can I get up and running?

Just say "when" – an authorized Avaya BusinessPartner can tailor a solution to your needs and your budget. By saving you money and helping you grow, IP Office repays your investment and lets you reallocate resources to other business priorities

Lowering long distance costs.

Routing telephone calls over the internet (Voice over IP or VoIP for short) – is growing in popularity. Particularly in the case of international calls, VoIP generates significant savings. If your company is already linking multiple offices using high-speed data circuits, the VoIP capabilities in IP Office make it possible to route voice calls over the existing infrastructure, providing another way to lower costs and leverage your investment. However you do it, the VoIP capabilities of IP Office are a way to put money back in your pocket.

Eliminating conferencing fees.

Collaborate with customers, partners, suppliers and remote employees. IP Office includes a 128 party audio conference bridge (64 max in one conference) to keep people working together and travel costs down. With Avaya IP Office, your organization can have its own private, secure conference bridge and eliminate expensive fees to third party conference providers.

Maximizing productivity

When employees can't get to the office (because of storms, personal issues or other reasons) IP Office allows them to work as at home (or any other location) as productively as though they were in the office.

And keep in mind...

IP Office delivers a whole range of capabilities. Only you can put a number on the value that many of these capabilities will have for your business.

Examples:

- Having calls automatically routed to a cell telephone or other location, so important customers can get through to the right person in real-time
- Being able to operate as a 24/7 business, without a 24/7 staff
- Using your communications to quickly identify when your top customers call

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How IP Office is benefiting businesses today.

- More room for sales
  With IP Office, a leading provider of commercial food service equipment now handles 50% more calls per day, without extra staff and without sacrificing the personal service it knows is the key to sales.

- At the head of the class
  By relying on IP Office to connect nearly 50 buildings, a public school system saved thousands of dollars on inter-office calls and simplified communications.

- Lowering global costs
  By using IP Office to hold teleconferences and make phone calls across the IP network, a strategic consulting firm is saving up to $30,000 per year.

IP Office can grow as your business grows to meet your needs.

Capacities: 2-384 extensions on a single location; up to 192 analog lines; 192/240 T1/E1 lines; 128 VoIP trunk lines; up to 1000 users across 32 locations

Call handling and messaging.

Get 24-hour support for callers/customers without a 24-hour staff. IP Office has a range of messaging, auto attendant and Interactive Voice Response (IVR) capabilities. Integrate messaging and advanced call handling into your customer service operations. Handle voice mail and email in a single mailbox.

Communication with customers.

Set up a formal or informal customer service center. Integrate your customer data base into your call handling. Manage the quality of your customer interactions.

Work anywhere.

Give your employees all the communications capabilities they have at the office whether they are working from home, a hotel or a remote office.

Collaborate with customers, vendors and remote staff.

Don’t pay any more fees to outside conferencing service providers. Use IP Office built in audio conferencing that is easy to set up and use while saving you money

Secure converged communications.

Use IP Office as a secure router with a built-in firewall/VPN. Route voice calls over a managed Internet service (VoIP) and pocket the savings. Simple administration Windows-based, graphical tools cut the time and expense of administration.

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Platform and Hardware Support

IP Office Control Unit Support

IP Office Release 7.0 is supported on the IP500 and IP500 V2 control units only.

Migration Opportunity for BCM and Norstar Customers / Users

For customers with a Nortel BCM or Norstar system Avaya IP Office now offers the possibility to upgrade the system to IP Office while keeping the existing Nortel telephones. IP Office Release 7.0 has been enabled to support these telephones and most of their features. Many features are the same as with BCM / Norstar and most of the rest are supported on IP Office but may have a slightly different look and feel.

This gives our BCM / Norstar customers the opportunity to have a smooth migration, as well as saving money, since the existing Nortel telephones can be mixed with Avaya IP Office telephones. Your investment is protected!

The following new hardware components of an IP Office system have been developed to support the Nortel telephones on IP Office.

TCM-8 Extension Card

Up to eight Nortel digital telephones can be connected to this 8-port extension card for IP500 and IP500 V2 control units. Up to four TCM8 cards can be plugged into an IP500 V2 control unit to support up to 32 digital telephones.

DS30A / DS16A Expansion Modules

For larger migrations from BCM or Norstar to IP Office two new IP Office Expansion Modules are available.

The DS16A supports up to 16 digital Nortel telephones and the DS30A up to 30. The DS16A is eqipped with one RJ21 connector, the DS30A with two of them.

IP Office Essential Edition - "Quick Mode" and "Standard Mode" Introduction

IP Office Essential Edition µ-Law and a-Law SD cards enable two new modes of operation – Quick Mode and Standard Mode.

When the IP500 V2 is first started, it boots-up in "Quick Mode", which offers a simplified management and e.g.: out of the box operations, two intercom buttons, basic system functionality, voice mailboxes on all telephones, sytem programming support using the Telephone User Interface (TUI), etc.

"Quick Mode" makes starting and using the system quick and easy.

Using Manager, users who need full IP Office Essential Edition functionality can simply change mode from "Quick Mode" to "Standard Mode". Once the system reboots, it re-starts in "Standard Mode" which is the way that IP Office Essential Edition work, looks and feels today.

Enhanced Offer for Small Systems with 20 Users or Less

The PARTNER Version, Norstar Version, and Essential Edition Quick Mode are specifically designed to address the "Sub-20" user market:

- Server-free operations – voice mail embedded in unit, eliminating the need for external servers.
- "Set it and Forget it" –system requires virtually no 'hands-on' support.
- Out of the box operations – automatically starts up in a basic telephone system mode. No programming required.
- Easy system management – system can be programmed using Telephone User Interface (TUI) commands or the Simplified Manager GUI software for more detailed administration.

Investment protection – users can start with the Quick Mode of operation, and switch to Essential Edition Standard Mode later without replacing hardware or licenses (excluding PARTNER ETR modules / ETR telephones).

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IP Office Essential Edition – PARTNER Version (North America)

PARTNER Version is designed to work directly out of the box without any programming or administration, allowing users to immediately make and place calls via analog trunks.

PARTNER Version provides the end user a similar desktop experience as the PARTNER ACS: e.g. two dedicated intercom buttons, key system functionality, Caller ID, etc. This helps minimize training requirements and provides the end user with the same comfort level they have come to expect with the PARTNER ACS system.

System Capacities

PARTNER Version supports larger station and trunk capacity as follows (may not be attained simultaneously):

100 Telephones:

- Up to 100 digital telephones (14xx series, 95xx series, Nortel T-series and M-series telephones)
- Up to 100 analog telephones
- Up to 18 PARTNER ETR telephones

64 Trunks:

- Up to 32 analog CO lines
- 1 PRI/T1 (24 lines)
- Up to 20 SIP lines

Investment / Migration protection – users can start with the PARTNER Version or Norstar Version, and migrate to Essential Edition without replacing hardware or licenses (excluding ETR modules / ETR telephones)

IP Office Essential Edition – Norstar Version (Middle East / North Africa only)

The IP Office Essential Edition – Norstar Version continues the evolution of communications systems for small businesses.

IP Office Essential Edition - Norstar Version is also designed to work directly out of the box without any programming or administration, allowing users to immediately make and place calls via analog trunks.

System Capacities

Norstar Version supports larger station and trunk capacity as follows (may not be attained simultaneously):

100 Telephones:

- Up to 100 digital telephones
- Up to 100 analog telephones

64 Trunks:

- Up to 32 analog CO lines
- Up to 12 BRI channels
- 1 PRI/E1 (30 lines)
- Up to 20 SIP lines

Investment / Migration protection – users can start with the Norstar Version, and migrate to Essential Edition without replacing hardware or licenses (excluding ETR modules / ETR telephones)

IP Office Release 7.0 adds support for the following telephone families:

- 9504 and 9508 digital telephones - 9600 series design, paperless keys, making it easy to change features and button assignments from remote
- 9608, 9621G, and 9641G IP Telephones - paperless keys, making it easy to change features and button assignments from remote. The 9621G and 9641G are equipped with a color touch display.
- BM12 LCD Button Module for the 9508, 9608, and 9641G – paper-less and with dual color LEDs

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9641G IP Telephone

9508 Digital Telephone

For Nortel BCM and Norstar customers IP Office now supports the following telephone series:

- T7000 Series Digital Telephones: T7000, T7100, T7208, T7316, T7316e (plus the T24KIM – Key Indicator Module)
- M7000 Series Digital Telephones: M7000, M7100, M7208, M7310, M7324 (plus the CAP 48 – Central Answering Position)
- Digital Mobility Solution 2,4 GHz:
  7420/7430/7440 Handsets, Base Station 10, Repeater 40, External Antenna 51, DMC 080, DMC 320
- Digital Mobility Solution 1,9 GHz (North America - 1929-1930 MHz):
  7439/7449 Handsets, Base Station 19, Repeater 49, External Antenna 51, DMC 080, DMC 320
- Digital Mobility Solution 1,8 GHz (Europe, Hong Kong, Australia, New Zealand - 1880-1900MHz):
  4135/4145/4145EX/4136/4146/4146EX Handsets, Base Station 15, Repeater 25/45, External Antenna 51

New Telephone Features

Support of the Nortel 1100 and 1200 IP telephones series was introduced with Release 6.1 with a basic feature set.
IP Office 7.0 adds new features to those telephones:

- “BCM-style” feature keys with central provisioning as well as e.g. status indication. Includes features like park/pickup, call forwarding etc. For details on the features refer to the product description
- Pick-up option for alerting Busy lamp keys
- Full management integration
- Access to IP Office directory (read only)

Enhancements to Public SIP Trunks

With IP Office 7.0 the following features for SIP trunks are introduced or enhanced:

- Programming templates
- Configurable parameters
- Multiple SIP Trunks
- Transparent Fax Over G.711
- Multiple Codes in LAN and WAN Calls
- Code Lock-Down
-

Mobility Enhancements

Switching of active calls between the office telephone and the twinning destination telephone is now simplified: by pressing a key on the desk telephone the call can be switched between the desk telephone and the mobile telephone back and forth.

When using one-X Portal for IP Office in telecommuter mode Mobile Twinning is also supported. A home office worker can now also leave their desk. Any incoming call will twinned to the mobile telephone and can be handled in the same way as in the office.

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Essential Edition Voicemail Enhancements

Essential Edition Embedded Voicemail comes with 2 ports and 15 hours as standard.
Additional ports can be added simultaneously increasing storage capacity

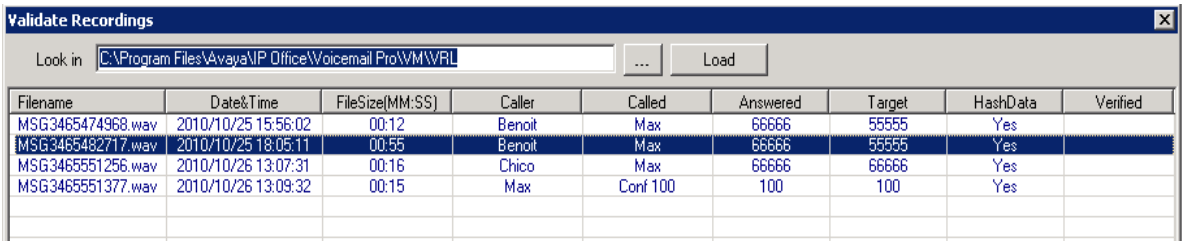
- 2 additional ports = 4 ports in total = 20 hours of storage
- 4 additional ports = 6 ports in total = 25 hours of storage

Further enhancements to Embedded Voicemail:

- Embedded Voicemail is providing an Automated Attendant and voice mail coverage activated for all telephones on Essential Edition.
- Embedded Voicemail Outcalling Notification – 1 number / 3 attempts
- Embedded Voicemail Message Alert Notification – displays which mailboxes (including Phantom Mailboxes) have new messages waiting for them

Preferred Edition Voicemail Enhancements

VoiceMail Pro Verified Recordings - VoiceMail Pro now has the option to deliver encoded recordings, that when presented back are verified if they have been unmodified.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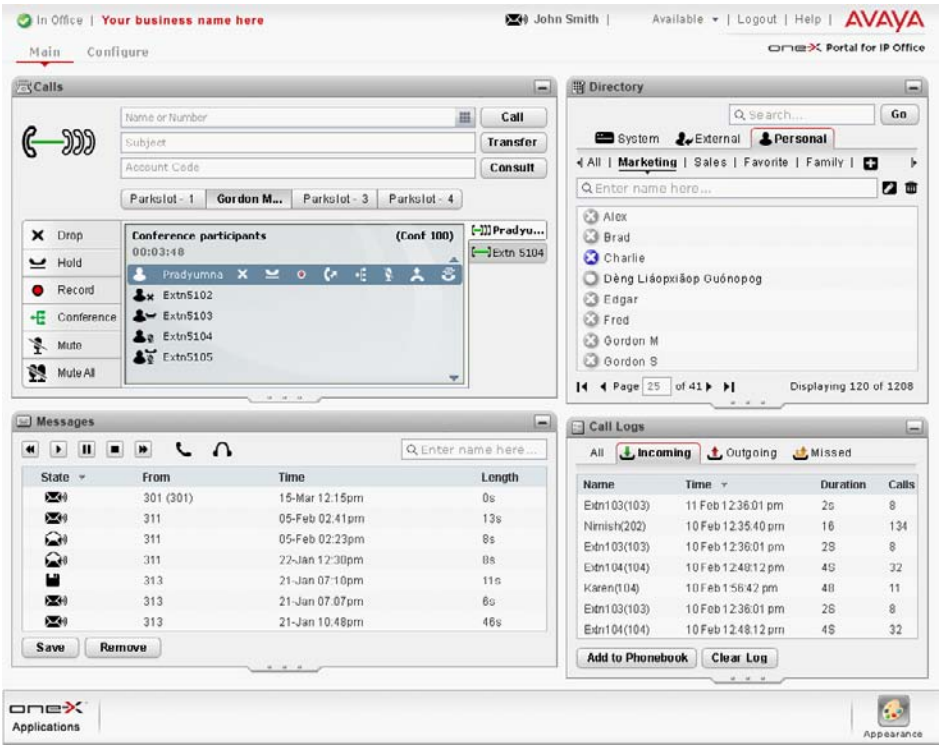
Load
Filename | Date&Time | FileSize(MM:SS) | Caller | Called | Answered | Target | HashData | Verified
MSG3465474968.wav | 2010/10/25 15:56:02 | 00:12 | Benoit | Max | 66666 | 55555 | Yes
MSG3465482717.wav | 2010/10/25 18:05:11 | 00:55 | Benoit | Max | 66666 | 55555 | Yes
MSG3465551256.wav | 2010/10/26 13:07:31 | 00:16 | Chico | Max | 66666 | 66666 | Yes
MSG3465551377.wav | 2010/10/26 13:09:32 | 00:15 | Max | Conf 100 | 100 | 100 | Yes

one-X Portal Enhancements

A new, modern, clean look and feel for the Avaya one-X Portal for IP Office web-client is introduced. The gadgets are now displayed in the Web browser with a 60/40 ratio, and can be resized, moved, minimized, and restored.



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Avaya one-X Portal for IP Office provides instant message chat sessions with other one-X Portal for IP Office users.

World Clock Gadget

The World Clock gadget displays the time in selected time zones that have been selected. It is possible to add up to 5 timezones to your world clock. Users can modify or delete them according to their needs.

The clock display can be viewed in either of two modes, Timezones or Clocks, selected by clicking on the appropriate tab.

Call Assistant

Avaya one-X Portal for IP Office supports a Call Assistant Plug-In which is a desktop application that installs on the user's PC (Microsoft Windows operating system only). Call Assistant can be used to receive incoming call notifications, answer calls, drop calls, launch the one-X Portal for IP Office browser, and configure screen options.

Customer Call Reporter Enhancements

CCR Agent Control Gadget in one-X Portal for IP Office

This gadget will provide the ability for enabled CCR Agents to define and control their agent presence status and hunt group membership.

It enables one-X Portal for IP Office to work as a CCR Agent with the functions:

- Login to a hunt group
- Logout from a hunt group
- Logout from all groups
- Change Agent State to busy n/a or ACW
- Support the input of reason codes

Historical Report UI Enhancements

The user interface in the supervisor screen for reports has been modified.

Upon selecting the Reports icon, a new reports page will be displayed. Along the top, there will be tabs for each of the report types available for CCR.

Further CCR Enhancements

Reporting:

- No printing on weekends option
- Option to save reports under different name instead of automatically overwriting

Wallboard:

- Logoff button
- Help button
- Animation Frame Rate setting – allows to reduce the frame rate SilverLight uses by default. This will reduce the CPU usage on the server if required

Customer Map:

- Help button

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Additional telephones supported on CCR statistics

- IP Office SIP Video Softphone
- Norstar™ digital telephones
- 1100 and 1200 SIP telephones

Monitor Statistics:

- All time statistics will be displayed in the HH:MM:SS format (following localization rules). This will allow the user to not have to figure out how many minutes and hours a statistic reported in seconds only in the past.
- Most averages and floating point statistics will be rounded to the nearest integer or the nearest 1 decimal point to increase clarity. This will apply to real time (Views, Dashboard and Wallboard) and historical statistics (reports). Keeping a decimal point for some statistics is required when it can affect competitiveness between agents or groups, hence affecting productivity.

Administration Enhancements

The IP Office Manager supports IP Office Essential Edition Norstar Version and IP Office Essential Edition PARTNER Version. In these versions, administrator can migrate from 2-digit dial plan to a 3-digit dial plan and vice-versa. The number of users and trunks is also increased from 48 to 100 users and up to 68 Trunks. The IP Office manager has been extended to support TCM8 card, DS30A/DS16A Expansion Modules and BCM / Norstar M-series and T-series telephones.

A flavor of IP Office Manager called the IP Office Simplified Manager now enables easy configuration of IP Office in IP Office Essential Edition Quick Mode. Quick Mode is managed by the Simplified Manager, a manager designed to reduce the installation times, specifically for Sub-20 type deployments. This is achieved by enabling configuration of features important for this market segment. The Quick Mode of operation further supports two sub-modes, namely KTS Quick Mode and PBX Quick Mode. Based on the type of SD card that is installed in the system, IP Office will boot up in one of the two Sub-Modes. IP Office always boots up in Quick Mode by default. An option to migrate/restart manager in IP Office Standard Mode is also provided. IP Office Manager Essential Edition Quick Mode offers task based configuration to speed up the installation process by providing easy and intuitive management.

The Data Migration Manager (DMM) is a migration tool designed to facilitate quick and easy migration from other products like BCM and Norstar to IP Office. The DMM tool is launched independently of the source and target applications. DMM will eliminate the need to re-record announcements and greetings when transitioning from BCM or Norstar to IP Office.

New Wall Mount

A new wall mount now offers enhanced cable management. Suitable for all IP Office control units and expansion modules all cables can be hidden under an attractively designed cover with cable management.

For customers who just need to wall mount the IP Office control unit the existing wall mount will remain in the IP Office portfolio.

Release License / Upgrade License

When upgrading to Release 7.0 from a previous release the IP Office 7.0 Release License must be present.

It comes in two flavors:

- Upgrade to Release 7.0 for systems with 32 extensions or less
- Upgrade to Release 7.0 for all systems with more than 32 extensions

For new installations it is not required to purchase one of these licenses.

90-Day Entitlement Period

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Beginning with IP Office Release 6.0 and continuing with IP Office Release 7.0 and the IPO500 V2 hardware, the system may be upgraded to the latest release of IP Office software (in this case Release 7.0) during the entitlement period; within 90 days after the system was first powered up by the customer. The customer has an entitlement period of 90 days from their initial use of the system to install the version of software they wish to run. Should another release of IP Office become available within 90 days of a system first being used, the customer may move to that release without purchasing an upgrade license. Example: if a customer first powered up the IP Office with Release 6.1, they are entitled to Release 7.0 within the 90 day period without charge for a Release 7.0 upgrade license. After the 90 day entitlement period, upgrade license charges will apply.

Voice Communication Solution Features

IP Office offers a comprehensive list of features and benefits for the small or mid-size business, including:

- Full PBX Features
  Caller ID, Call Forwarding, Conference Calling, Voice Messaging and more.
- Trunk Interfaces
  A variety of network trunk interfaces, including E1, T1, PRI, ISDN, SIP, analog loop start and analog ground start for comprehensive network connectivity. Not all trunk types are available in all territories, please check for local availability.
- Extensions
  Support for a range of extensions, from 2 to 384 that provide sophisticated voice performance for new and growing businesses.
- Telephones
  A variety of telephones including analog, digital and IP hard and softphones (wired and wireless) that provide the appropriate desktop or device telephone for every need.
- Advanced Call Routing
  Incoming calls are directed to the best available person or messaging service, according to the company's unique criteria.
- Alternate Call Routing
  Ensures reliable handling of calls by selecting from analog, digital or VoIP trunks.
- QSIG Networking
  Standards-based multi-site networking to interoperate with other PABXs (licensed feature).
- Integrated H.323 and SIP Gatekeeper and Gateway for Converged Communications
  The IP Office acts as an IP telephony server with Quality of Service (QoS) support through DiffServ for call routing
- SIP Trunking
  SIP trunking to Internet Telephony Service Providers. IP Office allows users with non-SIP telephones to make and receive SIP trunk calls

Data Communication Solution Features

For offices with basic data networking needs, IP Office can provide a complete data communications and networking solution:

- Internet Access
  Firewall protected leased line or dial-up connectivity via PRI, T1 or WAN port: high-speed dialed access, direct leased line connections for high usage and Web site hosting, integral security, and efficient access to information and a larger business presence via the Web.
- Routing
  Integral Static or Dynamic (RIP I/II) routing for both Internet and Branch-to-Branch solutions.
- Security
  NAT (Network Address Translation) and built in firewall to protect your internal network and IPSec support allows secure VPN data transmission across public IP Networks using 3DES encryption.
- DHCP
  Automatic IP address allocation for local and remotely attached PC's and other devices, including IP telephones.

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- LAN Switching
  The IP Office 500 V2 offers 2 switched Layer 3 Ethernet ports.

- LDAP Client Support
  For standards based directory synchronization with one-X Portal for IP Office and Phone Manager.

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Applications Platform Features

IP Office provides big business benefits and enhanced productivity for small and mid-size businesses with a full complement of sophisticated applications.

IP Office provides a number of free applications (eg. Phone Manager Lite, Microsoft TAPI service). These applications can be upgraded to provide enhanced functionality through the purchase of license keys. The complete list of applications are:

- Operator SoftConsole
  A graphical User Interface (GUI) for attendants on their PC desktop for call handling. Works with a telephone and is an easy way to learn and use sophisticated tools in a comfortable environment.
- Avaya one-X™ Portal for IP Office
  A powerful client/server web application that allows the user to control and manage phone calls from a web browser with the standard LDAP protocol for connectivity to external directories.
- Open CTI interfaces
  IP Office has a built in TAPI server that integrates easily with popular contact management applications such as Outlook. Sophisticated custom applications can be rapidly developed and deployed with our full software development kit.
- Voicemail
  Callers can always be answered with a personal voicemail greeting before a message is taken and message notification set. Messages can be shared (forwarded) with colleagues and retrieved by any telephone capable of tone dialing. When used with one-X Portal for IP Office, the PC can be used to control message playback.
- Integrated Voicemail to Email Presentation
  Voice messages can be copied into email messages and delivered into the email system. IP Office uses SMTP or MAPI* to deliver a copy of the voice message.

  Note: MAPI is not supported with VoiceMail Pro on Linux version delivered with the Application Server DVD
- Unified Messaging (UMS)
  The Unified Messaging Service within Preferred Edition provides a higher level of integration to synchronize both voicemail and email inboxes.
- Auto-Attendant
  Simplify service for administrators with this easy-to-use feature with the ability to construct customized automated services allowing callers to efficiently navigate the system, and reach the right person, without the assistance of an operator. Available with Preferred Edition and with Essential Edition.
- Interactive Voice Response (IVR) and Text to Speech
  Create automated customized systems allowing callers to interact with business information, for example, reading email, account enquiry systems, automated ordering systems, ticket purchasing systems, PIN checking, remote time sheet management, etc. Enhance these systems by using Text To Speech to read information back to callers. Available with Advanced Edition.
- Queue Manager and Campaign Manager
  Powerful voice and IVR applications for the Contact Center that facilitate agent and traffic management for better productivity and customer service.
- Customer Call Reporter (CCR)
  Available with Advanced Edition and combined with the built-in Automatic Call Distribution functionality of IP Office, CCR enables a small business to track and measure customer service and agent productivity levels
  - CCR delivers:
    - Simple/Intuitive Reporting
    - Browser/Thin Client Architecture
    - Single Server Efficiency (with Preferred Edition)
  - Target Segment: Small Business<30 agents
    - Can support up to 150 agents
    - 30 supervisors
    - One administrator

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Management Tools

Avaya IP Office is easily managed with IP Office Manager; a Windows based PC software application. It can be used from the LAN, remote on the WAN, or connected via the built-in RAS using a TA or a modem.

The System Status application is a useful diagnostic tool that provides enhanced details about equipment and resources in the IP Office system. This includes indication of alarms and details of current calls in progress for local or remote diagnostics.

Scalable Platform

The IP Office 500 V2 offers a modular, flexible chassis which with capacity for up to 16 analog trunks or 16 ISDN BRI trunks (32 channels) or 8 digital PRI trunks (up to 192 T1 channels or 240 E1 channels) using internal daughter cards. Up to 12 Expansion Modules may be added to provide a combination of up to 384 analog, digital or IP extensions, with additional analog trunks through external Analog 16 modules. Features include up to 148 optional voice compression channels, 2 independently switched LAN ports and built-in Embedded Voicemail.

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Telephone Options

IP Office supports multiple telephone solutions, giving the small and mid-size business maximum flexibility to choose according to their current and future needs.

- IP Telephones
  IP Office's integral H.323 server supports Avaya 1600 and 5600 series IP telephones, selected Avaya 9600 series IP telephones, Avaya T3 series IP telephones, Avaya 3600 series wireless VoIP telephones and IP Office Video Softphone.

- Digital Telephones
  IP500 digital station 8 cards and Combination Cards plus IP Office digital station 16 or 30 Expansion Modules support the Avaya 9500, 1400 and 5400 Series of digital telephones and Avaya T3 Series telephones, plus selected 2400 and 4400 Series telephones.
  The new TCM8 digital extension card as well as the DS16A and DS30A digital expansion module support Nortel T7000 series telephones and other telephones and button modules.

- ETR Telephones
  IP500 V2 ETR 6 cards support the Avaya ETR telephones as used on PARTNER ACS systems (Essential Edition – PARTNER Version only).

- Analog Telephones
  IP500 Phone 2, Phone 8 and Combination Cards plus IP Office Phone 16 or 30 Expansion Modules support standard analog telephones, faxes and modems, with support for calling line identification and message waiting indication where these services are provided.

- Wireless Telephones
  Avaya DECT R4 base stations can be added to support the Avaya DECT R4 3720 and 3725 telephones. The IP Office digital station interfaces support the Avaya 3810 telephone. The Avaya 3600 Series wireless VoIP telephones are also supported.
  With the new hardware modules TCM8, DS16A, and DS30A Nortel digital Mobility solutions are also supported.

- Third Party SIP Endpoints
  IP Office's integrated SIP Server supports third-party SIP endpoints such as, desktop telephones, softphones and conferencing speakerphones.

For a complete overview over supported telephones please refer to the Telephones 98 section of this Product Description.

Application and Feature Licensing

Some IP Office applications and features require a license key. Each license key is derived from the feature being enabled combined with the serial number of the Feature Key installed with the IP Office system.

IP500 and IP500 V2 systems use a mandatory Feature Key installed with every system.

Licenses are supplied in two forms; time limited trial licenses and indefinite licenses. Trial licenses allow applications to run in fully functional form for 60 days (from the date of license generation), after which time they cannot be used until upgraded at cost to the full license but can be ordered at any time during the product ownership.

For further details of the IP Office licenses including Trial Licenses refer to the Licenses 374 section.

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Chapter 2.
IP Office Essential Edition PARTNER® Version

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2. IP Office Essential Edition PARTNER® Version

The IP Office Essential Edition – PARTNER® Version (hereafter referred to as PARTNER Version) continues the evolution of communications systems for small businesses.

PARTNER Version is designed to work directly out of the box without any programming or administration, allowing users to immediately make and place calls via analog trunks.

PARTNER Version provides the end user a similar desktop experience as the PARTNER ACS: e.g. 2 dedicated intercom buttons, key system functionality, Caller ID, etc. This helps minimize training requirements and provides the end user with the same comfort level they have come to expect with the PARTNER ACS system.

PARTNER Version provides the bulk of the feature set that the PARTNER ACS R8 supported; plus integrates enhanced features from the IP Office Essential Edition platform, such as Mobile Twinning, Visual Voice Mail, 64-party Conferencing, SIP Trunking, Full PRI/T1 and more.

PARTNER Version provides the same 2-digit extension numbering plan as PARTNER ACS (default), and now supports a 3-digit flexible numbering plan (extensions 100 – 599).

PARTNER Version supports all current and "Euro" versions of PARTNER ACS telephones, the 1400 series digital telephones, the current 3920 and 3910 wireless products, as well as most analog devices (fax, wireless headsets, analog telephones). The original MLS series telephones and older wireless telephones are not supported.

PARTNER Version also supports a new line of Avaya digital telephones – the 9500 series telephones, and supports the Nortel T-series and M-series digital telephones.

PARTNER Version includes Embedded Voice Mail; providing a 2-port Automated Attendant (upgradeable to 6-ports) and voice mail coverage activated for all telephones. The Embedded Voicemail offers a variable number of hours of storage:

- 2-ports = 15 hours of storage
- 4-ports = 20 hours of storage
- 6-ports = 25 hours of storage

System Capacities

PARTNER Version supports larger telephone and trunk capacity as follows (may not be attained simultaneously):

100 Telephones:

- Up to 100 digital telephones (1400 series, 9500 series, Nortel T series and M series telephones)
- Up to 100 analog telephones
- Up to 18 PARTNER ETR telephones (additional non ETR telephones can be added total to reach system capacity)

64 Trunks:

- Up to 32 analog CO lines
- 1 PRI/T1 (24 lines)
- Up to 20 SIP lines

"Quick Mode / "Standard Mode"

IP Office Essential Edition µ-Law and a-Law SD cards include two new modes of operation – Quick Mode and Standard Mode.

When the IP500 V2 is first started, it will boot-up in "Quick Mode", which has the same look and feel as the PARTNER Version, e.g.: out of the box operations, 2 intercom buttons, key system functionality, voice mailboxes on all telephones, TUI programming support, etc. "Quick Mode" makes starting and using the system quick and easy.

Using Manager, users who need full IP Office Essential Edition functionality can simply Change Mode from "Quick Mode" to "Standard Mode". Once the system reboots, it re-starts in "Standard Mode" which is the way that IP Office Essential Edition work, looks and feels today.

System Programming

System programming is accomplished in 2 ways:

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- Telephone User Interface (TUI) from extension 10 or extension 11; and uses the same administration codes as the PARTNER ACS system.
- Graphical User Interface (GUI) – PARTNER Version provides a “Simplified Manager” mode in the IP Office Manager software; allowing you to program the system in less than 30 minutes

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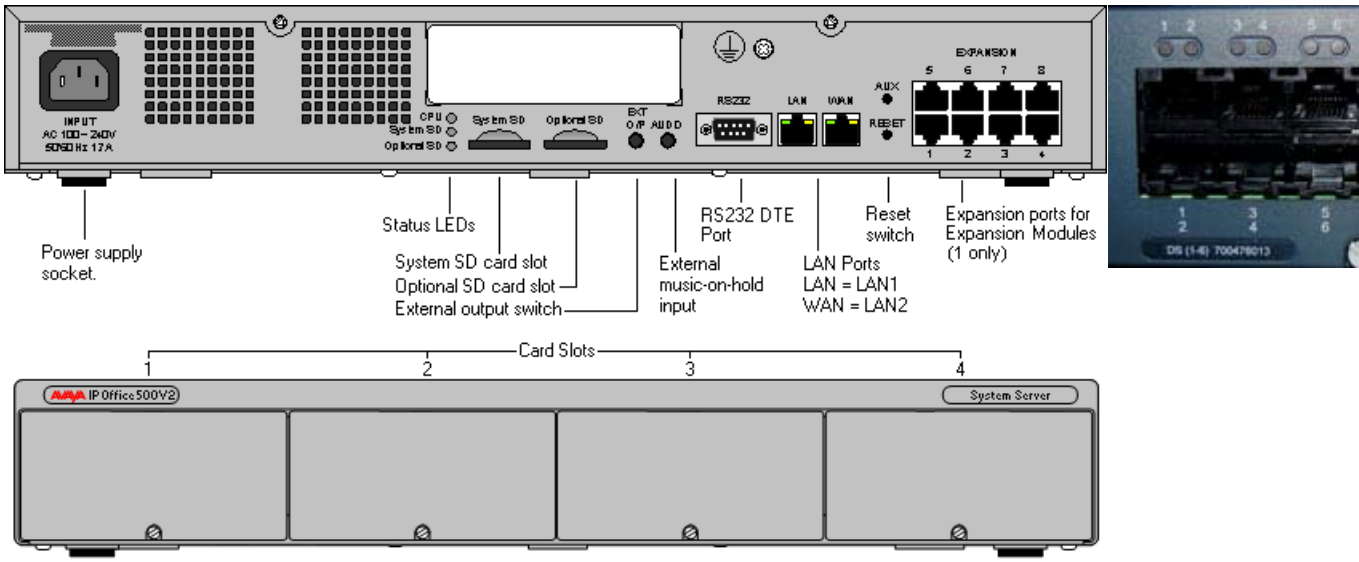
Control Unit

PARTNER Version software runs on the IP 500 V2 control unit. A PARTNER Version Secure Digital SD card or IP Office Essentia Edition µ-law Secure Digital SD card is required and is installed in the rear of the control unit, providing operating software, Feature Key, Licenses and directs the system to boot up in PARTNER mode or Quick Mode. An optional SD card may be installed in the optional SD card slot, providing redundant backup and system upgrade capabilities.



The front of the IP500 V2 Control Unit has 4 expansion bays, which supports the following system modules:

- Combination Card – Analog Trunks (Max 2 per system)
  The Combination Card provides 6 digital station (DS) ports (Ports 1-6); 2 analog station ports (Ports 7-8); 4 CO Line ports (Ports 9-12) and 10 Voice Compression Channels.
- ETR6 Card (Max 3 per system)
  The ETR6 Card provides 6 ETR (PARTNER ACS telephone) station ports (Ports 1-6); and supports (1) of the following optional trunk cards: ATM4 or PRI/T1. When an ATM4 is added, ports 7-8 become Power Failure Transfer ports, and ports 9-12 provide 4 additional CO Line Ports. Compatible only with PARTNER Version and µ-law SD cards, in North America market only.
- DS8 Card (Max 3 per system)
  The DS8 Card provides 8 digital station ports (Ports 1-8); and supports (1) of the following optional trunk cards: ATM4 or PRI/T1. When an ATM4 is added, ports 9-12 provide 4 additional CO Line Ports. This card supports only the 1400 and 9500 series telephones.
- TCM8 Card (Max 4 per system)
  The TCM8 Card provides 8 digital station ports (Ports 1-8); and supports (1) of the following optional trunk cards: ATM4 or PRI/T1. When an ATM4 is added, ports 9-12 provide 4 additional CO Line Ports. This card supports only Norstar® T-series and M-series telephones.
- Phone 2 (Max 4 per system)
  The Phone 2 module provides 2 additional analog station ports and supports one of the following optional trunk cards: ATM4 or PRI/T1.
- Phone 8 (Max 4 per system)
  The Phone 8 module provides 8 additional analog station ports and supports one of the following optional trunk cards: ATM4 or PRI/T1.

Expansion Modules

PARTNER Version supports following expansion modules:

- DS16 – provides additional 16 digital station ports (1400 and 9500 series telephones only)
- DS16A – provides additional 16 digital station ports (Norstar T-series and M-series telephones only)
- ATM16 (limit 1 per system) – provides additional 16 analog CO lines

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- Phone 16 – provides additional 16 analog stations
- Phone 30 – provides additional 30 analog stations
- DS30 – provides additional 30 digital station ports (1400 and 9500 series telephones only)
- DS30A – provides additional 30 digital station ports (Norstar T-series and M-series telephones only)

Trunk Cards

PARTNER Version supports the following trunk cards, which are mounted on any ETR6, DS8, TCM8, Phone 2 or Phone 8 base card:

- ATM4 – provides 4 analog CO line ports. Supports Caller ID
- PRI/T1 – supports full PRI or T1. Limit 1 PRI/T1 card per system

Telephones

PARTNER Version supports the following telephones:

- 1400 series digital telephones 30
- 9500 series digital telephones 31
- Norstar T-series and M-series digital telephones 31
- PARTNER ACS telephones 32
- PARTNER ACS "Euro" telephones 33
- Wireless telephones 34
- Analog "POTS" telephones 35

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1400 series telephones

The following 1400 series telephones are supported.

- 1416 Display Telephones
- 1408 Display Telephones
- 1403 Display Ttelephone
- DBM32 Button Module

Programmable Buttons

As well as the usual dialing keys, Avaya 1400 digital telephones have dedicated function buttons like Mute, Volume, Hold, Conference and Transfer. In addition to these, there are keys that can be programmed with a range of selected special functions. These keys can be used for calling other extensions on the system (Direct Station Select or DSS keys), or can be used for options from speed dialing numbers to controlling features such as Do Not Disturb. Many features use an indicator to show whether a feature is enabled.

Button programming is done through the TUI or Simplified Manager as part of the system configuration.

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9500 series telephones

The following 9500 series telephones are supported.

- 9504 Display Telephone
- 9508 Display Telephone
- BM12 Button Module

9504

9508

BM12

Programmable Buttons

The Avaya 9500 series telephones are “paper label free”, providing programmable labels in the display. The 9504 has 4 buttons (with Red & Green LED’s) and has 2 “pages” providing a total of 12 programmable buttons. The 9508 has 8 buttons (with Red and Green LED’s) and has 2 “pages” providing a total of 24 programmable buttons.

As well as the usual dialing keys, Avaya 9500 digital telephones have dedicated function buttons like Mute, Volume, Speaker, Headset, Call Logs and Contacts. In addition to these, there are keys that can be programmed with a range of selected special functions. These keys can be used for calling other extensions on the system (Direct Station Select or DSS keys), or can be used for options from speed dialing numbers to controlling features such as Do Not Disturb. Many features use an indicator to show whether a feature is enabled.

Button programming is done through the TUI or Simplified Manager as part of the system configuration.

Norstar T-Series and M-Series telephones

The following Norstar telephones are supported on PARTNER Version:

- T-Series Telephones: T7316E, T7208, T7100, T7000 and the T24KIM (not shown)

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- M-Series Telephones: M7324, M7310 and the CAP/KLM Button Module

PARTNER ACS telephones

The following PARTNER ACS telephones are supported on PARTNER Version:

- 6 Button Display Telephone
- 18 Button Display Telephone
- 34 Button Display Telephone (up to 4 per system, 2 per ETR card)

Not Supported:

- Call Assist-48 (CA-48) Adjunct

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Default Button Features:
When connected to the PARTNER Version, the top row of buttons on the 18D and 34D telephones come pre-programmed with the following features:

- Last Number
- Conference Drop
- Voice Mail Access
- Recall

PARTNER ACS "Euro" telephones

The following PARTNER ACS “Euro” telephones are supported on PARTNER Version:

- 6 Button non-Display Telephone
- 18 Button non-Display Telephone
- 18 Button Display Telephone
- 34 Button Display Telephone (up to 4 per system)

Not supported:

- Call Assist-24 (CA-24) Adjunct

Default Button Features:
When connected to the PARTNER Version, the top row of buttons on the 18D and 34D telephones come pre-programmed with the following features:

- Last Number
- Conference Drop
- Voice Mail Access
- Recall

Note: PARTNER ACS telephones are only available in the North America region, and work only with PARTNER Version and µ-Law SD cards running in “Quick Mode”.

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3920 Wireless

3910 Wireless

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T7406E Wireless

Digital Mobility 7400 Wireless

Note1: PARTNER ACS telephones are only available in the North America region, and work only with PARTNER Version and mu-Law SD cards running in “Quick Mode”.

Note 2: For further details on the telephones, please refer to the telephones chapter.

Analog "POTS" telephones

The system supports a wide variety of industry standard analog telephones and devices, for example: fax machines, credit card readers; wireless headsets, cordless telephones.

Analog devices are connected to the system:

- To one of the analog ports on the Combination Card
- Via a Phone 2 or Phone 8 Module
- To an ETR station port (directly or bridged with a PARTNER ETR telephone)

The major features work the same, regardless how they are connected: e.g. Hold, Conference, Transfer, Call Waiting, Message Waiting, Incoming Caller ID.

Analog telephones have access to all dial-code features familiar to PARTNER users: e.g. Call pickup, Line pickup, Call Forwarding activation/deactivation, etc.

Analog telephones connected to a phone module or Combination Card default to intercom dial tone. To place an outgoing call, the user will dial 9. Analog telephones connected to an ETR module follows programmed Automatic Line Selection.

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System Administration

The PARTNER Version and IP Office Essential Edition in Quick Mode can be programmed in 2 ways:

- Telephone User Interface (TUI) from extension 10 or extension 11; and uses the same administration codes as the PARTNER ACS system (see Table 1). Button Programming Features use the same codes as the PARTNER ACS system (see Table 2).

  TUI programming is an alternate way to do System Administration and Programming on the PARTNER Version. It is based on the PARTNER ACS TUI, but with improvements.

  Using TUI System Administration on ETR, 14xx, 95xx and Norstar sets, the buttons under the display become soft-keys which are used to navigate and change the administration items. There are also shortcut codes which enable the administrator to jump to particular administration items.

  Centralized Programming is similar to System Administration, but it is used to administer buttons and other options for any phone in the system.

  Individual Programming continues to be supported, allowing the phone user to program buttons on their sets.

- Graphical User Interface (GUI) – PARTNER Version provides a “Simplified Manager” mode in the IP Office Manager software; allowing you to program the system in less than 30 minutes.

  “Simplified Manager” is a PC GUI application inherent to the IP Office Manager application. Although system administration can be completed via either the TUI or GUI method; some advanced features such as SIP and PRI/T1 administration require the Manager application.

  When you execute Manager, the program searches for any IP500v2 units running PARTNER Version on your network and will automatically connect using the default login and password. You are then presented with a “Simplified Manager” Screen:

By using the System Hyperlinks, you are able to quickly and easily administer the PARTNER Version system in a short time. Further details can be found in the IP Office PARTNER Version Manager documentation guide posted on the Avaya Support website.

System Features

The system provides the majority of the feature set that Norstar and PARTNER ACS R8 supports. A listing of the built-in features is detailed below. Feature descriptions can be found on the following pages. TUI Feature Programming Codes (if applicable) are also provided. Yellow highlighted features are new in Release 7.0.

911 Emergency Dialling	Direct Line Pickup Features	Paging Features
Abbreviated Ringing	Display	PBX Mode
Absent Text	Display Language	Personal Line Termination

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Absent Text Message Inspect	Display SD Card Feature Key ID	Personalized Station Ringing
Account Codes	Display System IP Address	POT Support
Allowed/Disallowed Lists	Display System Release	Privacy
Application Programming Interface (TAPI)	Distinctive / Differentiated Ring	Recall
Attendant/Operator Positions	Do Not Disturb (Send All Calls)	Redialing Features
Auto Attendants - Emergency Greeting	Do Not Disturb Override	Remote Call Forwarding / Mobile Twinning
Auto Attendants - Multiple	Doorphones (max of 2)	Remote Access to Modem
Automatic Callback	Emergency Phone Number List	Ringing Line Preference
Automatic Daylight Savings Time	External Hotline	Ring on Transfer
Automatic Extension Privacy	Extension Hunt Groups	SD Card Startup / Shut Down via TUI
Auto Dialing	Extension Name Display	SMDR
Automatic Line Selection	Fax Machine Extensions	Speed Dial Features
Automatic VMS Coverage	Feature Button	Station Lock/Unlock
Backup Failure Alarm	Flexible Dial Plan	Sub-Menus (see Auto Attendants)
Backup Programming Automatic	Group Calling – Ring/Page	System Password
Backup Programming Manual	Group Hunting – Ring/Voice Signal	System Groups
Bridging (Joining Calls)	Group Pickup	Transfer / Transfer Return
Call Coverage	Hold	Transfer Return to Programmable Extension
Call Forwarding - Internal	Hold Release on Abandon	Unique Line Ringing with Override
Call Log (Digital Sets only)	Hold Reminder Tone	VMS Cover Button
Call Logs (ETR sets)	Hot Dial	VMS Cover Ring Interval
Call Pickup	Hotline	VMS Hunt Delay
Call Timer	Idle Intercom Timeout	VMS Hunt Schedule
Call Timer Continuity	Intercom Manual Dial	Voice Mail
Call Waiting	Intercom Dial Tone	Bypass Greeting

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Caller ID Features	Line Coverage Extension	Change Message Status to New
Cell Phone Connect (see Mobile Twinning)	Line Ringing Options	Choice of Language
Centralized Telephone Programming (from admin phones)	Message Alert Notification	End of Recording Options
CO Disconnect Time (Programmable)	Music On Hold	Outcalling Notification
Conference Drop	Mobile Twinning	Phantom Mailboxes
Contact Closure Support (2 devices)	Network Time Synchronization	Voice Mail to E-mail
Copy Settings	Night Service	Voice Mailbox Transfer – Direct
Direct Inward Dialing (DID)	On Hook Dialing	Voice Messaging
DID to Auto Attendant	One Touch Transfer	Visual Messaging
		Wake-up Service

911 Emergency Dialing

911 Emergency Dialing can be made from any extension, regardless of any restrictions.

Abbreviated Ringing (#305)

This feature activates or deactivates Abbreviated Ringing at the system telephone at a specific extension. When you are on a call and Abbreviated Ringing is Active, any incoming call rings only once. The green light next to the line button flashes until the call is answered or the caller hangs up (or for a transferred call, until the call returns to the transfer return extension).

This feature prevents incoming calls from distracting you when you are busy on another call. To allow calls to ring repeatedly, set Abbreviated Ringing to Not Active.

Absent Text Messaging (FEATURE 28)

This feature allows you to post a message (such as “Do Not Disturb” or “Away from desk”) on the display of your PARTNER ETR or digital system telephones. When another extension calls your extension, your active Absent Text Message appears on the caller’s display.

The system provides 15 pre-defined messages, plus 2 that may be customized by each user. Pre-defined messages include:

Back Soon	Back Tomorrow	Do Not Disturb
On Vacation	Out to Lunch	At Home
Away from Desk	Be Right Back	Please Call
On Holiday until	Meeting until	Don’t disturb until
With visitors until	With cust. til	At lunch until

Absent Text Message Inspect

The Inspect feature is a soft-key option, and allows you to remotely inspect the active Absent Text Message at another user’s extension.

Pressing the Absent Text Message button will provide you with an INSP option in the display. Press the button associated with the Inspect option and you are prompted to dial the extension number or press the Auto Intercom button you wish to view. When you do, the active message for that user is displayed.

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Account Code Entry (FEATURE 12)

Account Code Entry (ACE) is used to enter an account code (up to 15 digits) for an incoming or outgoing call currently being handled.

The Account code is printed in the SMDR record for the call for the customer to use for cost tracking purposes.

There are two different ways the feature can be used:

- Voluntary — Account Code Entry is optional.
- Forced – Account Code Entry is mandatory before dialing on an idle CO facility.

Forced account codes are validated against a list of pre-defined account codes which is created via the Manager application.

Allowed Lists(#407) / Disallowed Lists (#404)

Allowed Lists Use this feature to specify telephone numbers that users can dial regardless of other dialing restrictions, as long as they have access to an outside line. For example, if you restrict an entire category of calls through Disallowed Phone Number Lists (#404), you can permit calls to a specific number in that category by placing that number on an Allowed Phone Number List.

The System Administrator can create up to eight Allowed Phone Number Lists of up to 10 telephone numbers each.

Disallowed Lists allow you to specify telephone numbers that users cannot dial. For example, you may want to prevent calls to a specific telephone number or to categories of numbers, such as international numbers.

The System Administrator can create up to eight Disallowed Phone Number Lists of up to 10 telephone numbers each.

Allowed and Disallowed Phone Numbers can be up to 12 digits long and may include 0–9, #, *, and 'hold' (a wildcard character, displayed as "!").

Attendant Positions

Extension 10 and Extension 11 serve as system attendant positions. TUI System Administration can only be done at these extensions. Night Service and Unlock capabilities can only be initialized from Extension 10. When a user dials 0, Extension 10 will ring.

Auto Attendants (Multiple) / Sub-Menus / Emergency Greeting

The system supports up to 9 Automated Attendants / Sub-menus. A sub-menu is the same group of selector codes as is used in an Auto Attendant.

Each Auto Attendant has its own greeting (Morning / Afternoon / Evening), its own language (English / French / Spanish), its own set of selector codes (based on its time profile), and separate VMS Hunt and Delay Schedules.

Each Auto Attendant supports its own Emergency Greeting announcement, which can be remotely recorded and activated, and plays before the Auto Attendant Greeting.

Each Auto Attendant can be accessed by dialing the appropriate Auto Attendant dial codes (to record greetings, menu prompts, etc.).

Auto Dialing

Enables a user to dial outside numbers, extension numbers, feature codes or account codes with a single touch by pressing a programmed button.

3 types of auto dial are supported:

- Auto Dial – Outside
- Auto Dial – ICM (internal extension)

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- Auto Dial – ICM Page (voice signaling)

Automatic Callback

When calling an extension that is busy, press the Auto Callback softkey and the system will call you when the extension becomes free.

Automatic Daylight Savings Time (#126)

This feature automatically updates the system clock for annual Daylight Savings Time and Standard Time changes. Turn this feature off if you live in an area without daylight savings time.

Automatic Extension Privacy (#304)

Prevents other extensions with access to the same line from joining a call from their extension. Use this feature for extensions connected to a modem, fax, or any device whose function can be disrupted by someone trying to join it.

Automatic Line Selection

Specifies the order in which the system selects an available line (intercom or outside) when a user at the extension lifts the handset or presses SPEAKER to make a call without first selecting a specific line button.

For extensions with analog telephones, set Automatic Line Selection to intercom first. This enables analog telephones to access system features, including intercom calling. When users lift the handsets on analog telephones, they hear intercom dial tone. To access an outside line, they must dial 9.

Automatic Voice Mail Coverage (#310)

Use this feature to program a button to turn VMS Cover on and off, routing unanswered intercom and transferred calls for a users' extension to the Call Answer Service of the voice messaging system after the specified number of rings (factory setting is three rings).

Backup Failure Alarm

When an Automatic Backup fails, a 'Backup Failed' message is displayed on Extension 10 and 11.

Backup Programming - Automatic

The system does a nightly backup to the system SD card. No manual intervention or external PC is required.

Backup Programming - Manual (#124)

The administrator may do a manual system programming backup to the system SD card via the TUI.

Bridging (Joining Calls)

A user can bridge (join) on any active call by pressing the associated CO line button (assuming Privacy is not active on the originating extension.) When a call is bridged, the red and green LED's will alternate on all telephones which are bridged on the call.

Call Coverage (FEATURE 20)

This feature is used to redirect all Intercom, Transferred, DID and outside calls from a user's extension to another extension.

When Call Coverage is activated, covered calls are routed to the covering extension after a specified number of rings.

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Call Forwarding - Internal (FEATURE 11)

The Call Forwarding provides a means of redirecting Intercom (ICOM), CO, and Transfer calls from one extension to another specified extension.

Call Log - Digital Sets

Call Log is a fixed button on digital telephones and provides a visual record of calls made and received, including unanswered calls. Details are store for both users (maximum 30 entries) and hunt groups (maximum 10 entries). The method of operation varies according to the phone type but in all cases the call records can be used for return calls.

Call Logs - ETR sets (#317)

Use this feature to program extensions to log answered Caller ID calls so that calls answered at an individual extension can be viewed in the Caller ID Call Log. Once the feature is activated for an extension, when a user activates the Caller ID Call Logging and Dialing (F23), all Caller ID calls that are answered by that extension are logged.

Up to 3 extensions can be designated to log all calls, both answered and unanswered, regardless of where they terminated in the system.

Call Pickup (INTERCOM 6xx)

This feature is used to answer any intercom, transferred or outside call ringing at another specified extension.

Call Timer Display

Displays the duration of a call from the time it was answered.

Call Waiting

Call waiting applies to analog, single line telephones, and applies to the following types of calls:

- Intercom calls
- Transferred calls
- Forwarded calls
- Covered calls
- Outside calls (if the extension has a line appearance of the line)
- Voice Signaled calls

Caller ID Features

The following Caller ID features are supported:

- Caller ID Inspect (FEATURE 17)
  Allows you to view Caller ID information for a second call, without disconnecting the current call or putting it on hold. Use this feature to inspect ringing, active, or held calls.
- Caller ID Name Display (FEATURE 16)
  When an incoming call is ringing at a display telephone, both the name and number appear on the user's display. The Number is displayed on the top line, and the Name on the 2nd line. In some cases users may wish to see the Caller ID Name on the top line and the Caller ID Number on the 2nd line. This feature provides this capability.

Cell Phone Connect

See Mobile Twinning 48.

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CO Disconnect Time (Programmable)

Use this feature to change the hold disconnect time for an outside line. Different telephone companies use different length signals. The length of the signal is called the CO or Hold Disconnect time. If you place a call on hold and the caller hangs up, but the call does not disconnect within a minute after the caller.

Conference Drop

On 1400 and 9500 digital telephones, you may choose which specific party you want to drop from a conference call. On ETR and analog telephones, the last party added to the conference call is automatically dropped.

Conferencing

The system supports up to 64-party conferencing. A maximum of 2 analog lines can be in any conference call. Conference calls may consist of combinations of analog lines, digital lines (PRI/T1, SIP), analog telephones, ETR telephones or digital telephones.

Contact Closure Support for 2 Devices (FEATURE 41 and FEATURE 42)

The IP500 V2 Control Unit has a Contact Closure control jack on the back panel. This is a 3.5 mm stereo jack that can drive two external adjunct relays.

This feature allows users at eligible extensions to control an external adjunct by entering feature codes at their system or standard telephone. The port can control a variety of devices, the most common being door strikes, night bells, lighting, or voice recording systems.

Copy Settings (#399)

This feature allows a System Administrator to copy extension-specific programming from one extension to another. When the Extension Copy operation is performed, the destination extension will have the same properties as the originating extension.

Extension Copy can only be performed from an Attendant Extension in System Programming Mode or from the Windows Manager PC application.

Dial Code Features

- 'F' represents the FEATURE button.
- 'I' represents the INTERCOM button.
- 'XX' represents a system extension number.
- 'PP' represents a Personal Speed Dial code.
- 'G' represents a single-digit group number.
- 'LL' represents a system line number.

Feature	Partner Version Code	Norstar/BCM Code
Do Not Disturb	F01	F85
Do Not Disturb Cancel	F01	F#85
Recall	F03	F71
Save Number Redial	F04	F67
Last Number Redial	F05	F5
Conference Drop	F06	F934 (new)
Privacy	F07	F83
Call Forward	F11	F4
Call Forward Cancel	F11	F#4

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Account Code Entry	F12	F900
Voice Mailbox Transfer	F14	F986
VMS Cover	F15	F984
Caller ID Name Display	F16	F933 (new)
Caller ID Inspect	F17	F811
Call Coverage	F20	F932 (new)
Station Lock	F21	F936 (new)
Station Unlock	F22	F937 (new)
Caller ID Log	F23	F812
Hot Dial	F26	F935 (new)
Message Alert Notification	F27	F938 (new)
Absent Message	F28	F931 (new)
Contact Closure 1	F41	F9*41
Contact Closure 2	F42	F9*42
System Release	F590	F9*80
System IP Address	F591	F9*81
SD Card Serial Number	F592	F9*82
System Speed Dial	F6XX (00-99)	F06XX (00-99)
Personal Speed Dial	FXX (80-99)	F*4XX (80-99)
DND Override Numbers	F7XX (00-19)	F07XX (00-19)
Call Transfer	Fixed Button	F70
Voice Announce to Extension	I* + XX	F66 + XX (new)
Call Park to own Extension	Transfer + XX	F74 (new)

Direct Inward Dialing (DID)

Use this feature to have Direct Inward Dialing (DID) numbers on a PRI / T1 or SIP line routed automatically to a specific extension, hunt group, or calling group. When a Direct Inward Dialing (DID) call arrives, the PARTNER system “collects” the digits from the T1 service provider (that is, the number that was dialed) and matches the number to a DID Mapping Table you created. If the incoming number matches an entry in the DID Mapping table, the call rings at the matching extension or hunt group.

Direct Inward Dialling (DID) to Auto Attendant

DID calls can be directed to and answered by any Automated Attendant.

Direct Line Pickup (INTERCOM 68LL, INTERCOM 8LL)

With the Direct Line Pickup features, you can access a ringing or held call or a call in progress. You can also select a line to use that is not assigned to your telephone. The Direct Line Pickup features are:

- Direct Line Pickup—Active Line
- Direct Line Pickup—Idle Line

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Display Language

All system telephones offer the choice of 3 display language:

- US English
- Canadian French
- Latin American Spanish

Each individual extension may select their own language.

Display SD Card Feature Key ID (F592)

Displays the SD Card Feature Key ID on any display telephone.

Display System IP Address (F591)

Displays the IP address currently assigned to the IP500 V2 system.

Display System Release (F590)

Displays the software release running on the IP500 V2 system.

Distinctive / Differentiated Ringing

The system provides different ring patterns for external calls, internal calls and transferred calls.

Do Not Disturb (FEATURE 01)

Use this feature to be able to press a programmed button to prevent incoming calls for the extension from ringing (lights still flash). When Do Not Disturb is on, outside callers hear ringing while inside callers hear a busy signal. Calls are immediately sent to your voice mailbox.

Do Not Disturb Override

Allows a user to input a list of up to 20 telephone numbers and/or internal extension numbers that will ring at the users' station when Do Not Disturb is activated.

Doorphones

A doorphone is usually placed near an entrance, to screen visitors. You can connect up to two doorphones to the system.

Emergency Phone Number List (#406)

Create a list of up to 10 telephone numbers that all users can dial regardless of dialing restrictions, provided they have access to an outside line. Typical list entries include fire, police, and emergency services numbers. police, and emergency services numbers.

Extension Hunt Groups (#505)

Use this feature to assign any number of extensions to a Hunt Group. When extensions are in a Hunt Group, an incoming call searches or "hunts" for the longest idle extension that is available. The system supports up to six Hunt Groups

Extension Name Display

The user' name and extension number is displayed on the phone display.

External Hotline (#311)

When a user lifts the handset of an external hotline, a predetermined outside number is dialed automatically. The external number might be, for example, a frequently called service bureau. The external hotline must be a single line telephone, not a system telephone, and should not have a dial pad.

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Fax Machine Extensions (#601)

Allows you to connect your fax machine to your system, allowing you to share network facilities.

Flexible Dial Plan

Through Manager, the system defaults to a fixed 2-digit dial plan (extensions 10 – 57); but can be changed to a flexible 3-digit dial plan (extensions 100 – 599).

Group Calling - Ring / Page (INTERCOM 7G / INTERCOM *7G)

Use this feature to ring, page, or transfer calls simultaneously to all of the extensions in any one of four Calling Groups. (G represents a Calling Group number from 1–4.) When paging, you hear a beep and can begin speaking. Your voice is heard on the speakers of all idle system telephones in the selected Calling Group. The first extension to answer the call (on ETR sets - by picking up the handset or pressing SPEAKER, on 14xx sets - or by pressing the ANSWER softkey on the display) is connected to the caller.

You can group page employees for general announcements; this is an inexpensive alternative to a paging system.

Group Hunting - Ring / Page (INTERCOM 77G / INTERCOM *77G)

Use this feature to distribute call volume among extensions in a group, to off-load call activity from a single user. When an intercom or transferred call is placed to a Hunt Group (where G is a Hunt Group number from 1–6), the system rings or voice-signals the first available extension in the group, passing over busy extensions (or those with Do Not Disturb on) in a circular hunt. If a ringing call is not answered within three rings, the call moves to the next available extension, and so on, until the call is answered or the caller hangs up.

For a voice-signaled call, only the first available extension is signaled; if that extension does not answer, the call does not keep hunting. After an extension receives a Hunt Group call, the next call to the Hunt Group will not ring or voice-signal that extension first unless all other extensions in the group are busy or do not answer.

Group Pickup (INTERCOM 66G)

Use this feature to answer any outside, intercom, or transferred call ringing at an extension in a Pickup Group (where G is a Pickup Group number from 1–4). When a call rings at an extension that is in a Pickup Group, you can answer the call at any extension without knowing which extension or line is ringing and without being in the same Pickup Group. The system can have up to four Pickup Groups.

Hold Reminder

If a call is left on hold for longer than a specified period, the extension will ring back, alerting you that the call is still on hold. This ring continues to sound until the held call is retrieved, or until the caller hangs up.

Hot Dial

This feature enables a user to start dialing a number without first pressing the speaker button or going off-hook.

Hotline

This feature allows designated extensions to automatically place an intercom call to another designated extension, when it goes off-hook.

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Intercom Dial Codes

- 'I' represents the INTERCOM button.
- 'XX' represents a system extension number.
- 'G' represents a single-digit group number.
- 'LL' represents a system line number.

Intercom Dial Code	Feature
I6XX	Call Pickup
I66G	Group Pickup
I68LL	Direct Line Pickup—Active Line
I70	Loudspeaker Paging
I*70	Simultaneous Paging
I7G	Group Calling—Ring
I*7G	Group Calling—Page
I77G	Group Hunting—Ring
I*77G	Group Hunting—Voice Signal
I8LL	Direct Line Pickup—Idle Line

Intercom Dial Tone (#309)

Use this feature to determine the type of dial tone that the system provides at an extension. It may be necessary to change this setting to Machine (outside line) dial tone for an autodialing device, such as a fax or modem, that has trouble making calls. For example, if you have a modem that checks for outside line dial tone before dialing, use this procedure to change from Regular to Machine dial tone.

Intercom Manual Dialing

Users can make an internal call to another extension by ringing it (dialing its extension number) or voice signaling the extension by prefixing the extension number with a *.

Line Coverage Extension (#208)

Use this feature to identify an extension as the "owner" of a specific outside line, so the extension can activate Call Coverage or VMS Cover for that line.

Line Ringing Options

This feature specifies the options: Immediate Ring, Delayed Ring or No Ring for incoming calls on a line appearance programmed on the extension.

Log All Calls

Up to 3 extensions can be designated to log all calls, both answered and unanswered, regardless of where they terminated in the system.

Message Alert Notification (FEATURE 27)

Message Alert Notification allows any ETR or Digital telephone, programmed with Auto Intercom buttons, to see which users have new messages waiting in their mailbox.

When the Message Alert Notification button is pressed, the DSS/BLF buttons programmed on the user's telephone will display a flashing red LED when the associated user has new voice mails waiting.

This feature may be left on at all times or deactivated by pressing the Message Alert button again.

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Music on Hold

Connect an audio source to the Music-On-Hold port on the processor module so that a caller hears music when placed on hold when the Music-On-Hold feature has been activated.

Network Time Synchronization (#128)

Use this feature to specify whether you want to synchronize the system clock with the network time delivered by your service provider over Caller ID. If Network Time Synchronization is set to Active, the system will use the network time provided by your service provider; if it is set to Not Active, the system will use the time provided by its system clock.

Night Service

When Night Service is on and a call comes in, all extensions in the Night Service Group ring immediately, regardless of normal Line Ringing settings.

One-Touch Transfer

This feature allows a user to automatically transfer a connected call with one-touch -- by pressing a pre-programmed Auto Dial Intercom for the destination extension. (The *Transfer* button is not used.)

- The transfer is completed when the transferring station goes on-hook, or touches the *Transfer* button or *Complete* soft-key.
- The user initiating the transfer is able to consult and complete the transfer or blind transfer after successfully initiating a one-touch transfer.

The LEDs at the phone initiating the transfer, as well as other appearances of the call in the system, behave the same as if the user touched the *Transfer* button to initiate the transfer.

Paging Features (INTERCOM 70 and INTERCOM *70)

- Loudspeaker Paging - Users can initiate a call (either manually dialed or intercom auto dialed) to an extension connected to an analog station port that connects them to an external loudspeaker paging system.
- Simultaneous Paging is a feature that pages both the external loudspeaker paging system (via the paging port) as well as voice signals all of the extensions in Calling Group 1.

PBX Mode

The system can operate in Key System mode (default) where all lines appear on all telephones; or can be administered to operate in PBX-mode. PBX-mode provides 3 Intercom buttons per telephone which work similar to System Access buttons (provides intercom dial tone, d1al 9 to access an outside line).

Note: On PARTNER ETR sets, the system provides 2 Intercom buttons per telephone.

Personal Line Termination

Allows you to assign a line / lines to specific extension / extensions.

Personalized Station Ringing (#323)

Allows you to assign a personalized ring pattern to an extension

POTS (Plain Old Telephone Set) Support

POTS (single line analog telephones) are supported on the system. They are connected to an analog station port on the Combo module; a port on the Phone-2, Phone-8 or Phone-16 module, or to an ETR port.

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Privacy (FEATURE 07)

Privacy prevents other users from joining on your call.

Recall (FEATURE 03)

Use this feature to send a timed switch hook flash over the telephone line to “recall” a new dial tone or to access certain central office features, such as Call Waiting or 3-way calling.

Redialing Features

There are 2 features which offer redial functionality:

There are 2 features which offer redial functionality:

- Last Number Redial (FEATURE 05): Last Number Redial redials all digits dialed on the last outside call except account codes.
    - On 1400 series telephones, there is a fixed Redial button, which allows you to scroll through and redial the last 20 number called from that telephone set..
- Save Number Redial (FEATURE 04): Saves into temporary memory the last outside number (up to 28 digits) dialed from a system telephone. Use this feature to save a number before you hang up on a busy or unanswered call. Once saved, the number can be redialed at any time. The number stays in memory until a different one is saved.

Remote Call Forwarding / Mobile Twinninng

There are two types of Remote Call Forwarding supported on PARTNER Version:

1. Remote Call Forwarding (FEATURE 11) allows a user to send internal and external calls destined for a particular extension to an external phone number (off the system) over an analog trunk. Remote Call Forwarding will not alert the internal extension, but immediately sends the call out to the predefined number.
2. Mobile Twinning allows a user to send internal and external calls destined for a particular extension to an external phone. The call simultaneously rings at the internal extension and receives the extension’s coverage (including Voicemail) treatment if the call has not been answered. This feature is only available on systems that have digital trunks (SIP, PRI/T1).

Remote Access to the Built-in Modem

This feature allows for unattended access to the built-in modem (x76) by calling into the system:

- By a DID number, or
- By transferring to the modem via the Automated Attendant.

When an Automated Attendant Selector code with the "Transfer To Number" action has the modem as its destination, the system will transfer the call to the built-in modem.

Ring on Transfer (#119)

This is used to specify what the caller hears while they are being transferred. If Ring on Transfer is set to Active, callers hear ringing while they are being transferred; if it is set to Not Active callers hear beeps or Music on Hold if this is activated and a music source is connected to the system.

Ringing Line Preference

When an extension is on-hook and the extension is ringing, the user simply goes off-hook to answer the call and is automatically connected to the ringing call. If more than one call is alerting, the system automatically connects the user to the longest ringing call.

SD Card Start-up / Shut Down via TUI (#733)

A new TUI command is provided to shut down and restart the system SD card slot and the optional SD card slot, without shutting down the entire system. This allows for removal and reinstallation of the system SD and optional SD cards.

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Speed Dial Options

The system supports the following types of Speed Dialing:

- Personal Directory - 1400 and 9500 Telephones only
  Each user can store up to 20 frequently-dialed names and numbers in their extension. Personal Directory numbers can be dialed quickly by pressing CONTACTS and dialing the first few letters of the person's name. Personal Directory numbers programmed for a particular extension are for use only at that extension.
- Personal Speed Dial (FEATURE 80 – 99) – ETR and Analog Telephones only.
  Each user can store up to 20 frequently-dialed numbers (no name) in their extension. Personal Speed Dial numbers can be dialed quickly by pressing FEATURE (or # at intercom dial tone on a single-line telephone) and the two-digit speed dial code from 80 through 99. The Personal Speed Dial numbers programmed for a particular extension are for use only at that extension.

  On analog telephones, a user will press the # key (instead of the Feature key) then dial the 2-digit number.
- System Speed Dial – 1400 and 9500 Telephones only
  System Speed Dialing is a shared list of up to 100 frequently-dialed names and numbers (FEATURE 600 – 699). All users on the system may use these numbers by pressing CONTACTS and dialing the first few letters of the party's name.
- System Speed Dial (FEATURE 600 – 699) – ETR and Analog Telephones only.
  System Speed Dialing is a shared list of up to 100 frequently-dialed numbers (no name) up to 28 characters in length. All users on the system may dial a System Speed Dial number by pressing FEATURE (or # at intercom dial tone on a single-line telephone) and the three digit speed dial code, from 600 through 699

Station Lock / Unlock (FEATURE 21)

Station Lock helps to prevent unauthorized people from making outside calls from extensions. Users enter a four-digit code on their telephone dialpad to "lock" your extension. To unlock the phone, the user enters the identical code.

Station Message Detail Reporting (SMDR)

SMDR is a call reporting feature that provides records of call activity. It is commonly used in many types of business, including Legal, Contact Centers, Sales and Real Estate. Call reporting information allows users to:

- Detect any unauthorized calls.
- Bill clients or projects.
- Bill back by department.
- Reduce telephone costs by identifying the need to change telecommunications services
- Print Caller ID information.

The output is generally sent to a PC running an optional Call Accounting software package.

System Groups

PARTNER Version supports the following types of groups:

- Pickup Groups (4 Groups) (INTERCOM 6xx)
  When a call rings at an extension assigned to a Pickup Group, a user at any other extension in the system can answer the ringing call by dialing the Pickup Group code. The Pickup Group feature helps when a user needs to answer calls on lines or pools not assigned to his or her telephone.
- Calling Groups (4 Groups) (INTERCOM 7x / *7x)
  A Calling Group is a group of extensions that can be called at the same time. Any user in the system can ring or page all extensions in a Calling Group at the same time or transfer a call to a Calling Group. The first extension to pick up the call is connected to the caller. A typical use of this feature is to have callers ring into a Calling Group of sales representatives, or to create a "Page All" group.
- Hunt Groups (6 Groups) (INTERCOM 77G / *77G)
  When extensions are in a Hunt Group, an incoming call searches or "hunts" for the first available extension.
- Night Service Group (1 Group) (#504)
  When Night Service is activated and a call comes in, all extensions assigned to the Night Service Group ring immediately, regardless of normal Line Ringing settings.

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System Password (#403)

Allows you to define a four-digit password that users can enter from system telephones to override dialing restrictions (if the extension has access to an outside line) or turn Night Service on and off. Not available on analog telephones.

Transfer / Transfer Return

You can transfer calls to an extension or group using the TRANSFER button on a system telephone or the switchhook on a single-line telephone.

Transferred calls return to the originating extension if they are not answered. You also can program the number of times a transferred call rings before it returns to an extension.

Transfer Return to a Programmable Extension (#306)

This feature provides an option to re-route unanswered transferred calls to an alternate extension.

- Programmable on a per extension basis.
- The existing System Programming option, *Transfer Return Rings*, will be used to indicate when transferred calls are returned to the Transfer Return extension.

If a call is routed to a transfer return extension and there are no available intercom appearances to terminate the call, the call will continue to alert at the transfer destination until the transfer return extension becomes available.

The default value for each transfer return extension is its own extension.

Unique Line Ringing (#209) / Override Line Ringing (#209)

This feature allows users to differentiate by sound which line is ringing by assigning a ring pattern to it. Once a ring pattern is assigned to a line, incoming calls on that line ring with the assigned ring pattern.

- 8 ring patterns are available.
- Unique Line Ringing does not apply to DID calls because the channel used is not guaranteed.
- The default is pattern 1.

If the Override Line Ringing feature is active, then incoming calls on a line that ring at an extension ring with the configured extension ring pattern.

Visual Voice Messaging

Supported on 1400 and 9500 digital telephones only. Allows the user to manage their voice mail messages via their display. Features supported include:

- access new /old / saved messages.
- next and previous message.
- fast forward and rewind.
- pause message.
- save, delete and copy message to other users of the system.
- change default greeting.
- change password.
- change message status from old / saved to New.

Voice Mail Options

- Bypass Greeting
  - After a call goes to an extension's voice mailbox, the caller can press the 1 key to bypass the mailbox greeting and begin recording a message.
- Change Message Status to New (*06)
  - This feature provides the ability to change the status of a message from *Old* or *Saved* to *New*.
  - Users can dial *06 at any time after the *Old* or *Saved* message begins to play through the 3 second pause after the caller information portion of the message.

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        - The message is moved to the queue of New messages.
    - When a message is marked as *New*, the system does not initiate a new Voicemail to Email message, but the extension's message waiting light is illuminated.
- Choice of Language (English / French / Spanish)
    - Each Auto Attendant can have a primary language assigned to it. The language applies to the pre-recorded greetings and prompts available to that Auto Attendant.
    - This overrides the system language setting.
    - The per-extension Display Language setting will also override the system language and change all the displays, pre-recorded greetings, and prompts to the chosen language.
- End of Recording options
    - When a caller has finished leaving a message in an extension's mailbox, the caller can touch the # key to signal the end of recording of the message.
    - If the message is less than 3 seconds, the system will play the following prompt: "Message too short. Deleted. Goodbye."
    - If the message is longer than 3 seconds, the system will play the following prompt: "Your message has been sent. Goodbye."
- Outcalling Notification
    - If you use the Outcalling option, Embedded Messaging will call the number which you have programmed to signal that a new message has arrived in your mailbox.
        - Each user can have 1 number programmed for outcalling notification.
        - The system will make up to 3 attempts (in 5-minute intervals) to contact the user.
        - After 3 attempts, outcalling notification will cease until another new voice message arrives.
- Phantom Mailboxes
    - Provides voice mailbox support for all extensions (10-57), without requiring the physical hardware to be present. Extensions without hardware are called *phantom extensions*.
        - Note: Ports 7 and 8 of an ETR-6 module can be used for phantom extensions even though you cannot plug a phone into them.
    - Calls to a phantom extension go directly to voice mail.
        - DTMF breakout service can be used to transfer the call.
    - Phantom voice mailboxes:
        - Default like a normal mailbox.
        - Can be accessed remotely if the remote access feature of the mailbox is enabled.
        - Voicemail to Email can be activated for a phantom mailbox via the Simplified Manager.

VMS Cover (FEATURE 15)

Allows you to turn VMS Cover on and off, routing unanswered intercom and transferred calls for users extensions to the Call Answer Service of the voice messaging system after the specified number of rings.

VMS Cover Ring Interval (#321)

Allows you to define the number of times a call rings before it is sent to a user's voice mailbox.

VMS Hunt Delay (#506)

Allows you to determine when outside calls should be answered by the Automated Attendant of the voice messaging system. You can set the system for any number of rings, 0–6. Assigning more rings gives the operator an opportunity to answer calls before they go to the Automated Attendant.

VMS Hunt Delay is programmable on a per-line basis. In addition, you can program this feature so that calls can be handled one way during the day and a different way when the system is in Night Service.

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VMS Hunt Schedule Interval (#507)

Allows you to determine when outside lines are covered by the Automated Attendant. Options include all the time, only during Day operation (Night Service is off), or only during Night operation (Night Service is on).

Voice Mailbox Transfer - Direct (FEATURE 14)

Allows you to transfer a call directly into a users voice mailbox without ringing their extension.

Voice Messaging

The PARTNER Version has Embedded Voicemail included in the system. All extensions are automatically assigned a mailbox.

Embedded Voicemail also provides:

- A 2-port, single level Automated Attendant
- Separate Morning, Afternoon, Evening and Out of Hours menu greetings with time profiles
- Dial by Name capability
- Variable hours of storage:
  - 2-ports = 15 Hours
  - 4-ports = 20 Hours
  - 6-ports = 25 Hours
- Up to 3 minutes per message
- Fast Forward, Rewind, Replay and Skip capability
- Voice Mail to e-mail capability
- Remote message retrieval
- Visual Messaging (on digital telephones only)

Wake-Up Service

This feature allows a system phone (x10) to set a Wake-up Call on behalf of another user.

- When a Wake-up Call is scheduled, the system will place an intercom call to the target extension at the scheduled time.
  - Wake Up calls ring for approximately 30 seconds (and overrides DND).
  - If the target extension is busy on a call or has an alerting call, the Wake Up call is delivered to the phone as an intercom call.
  - If the Wake Up call is answered, the user will hear Music On Hold if it is active; otherwise, they will hear silence.
  - If the Wake Up call is not answered, the system will make a 2nd attempt 5 minutes later.
    - If the call is not answered after the 2nd attempt, then the Wake Up call is abandoned.
  - A scheduled Wake Up call occurs once in a 24 hour period.

Application Programming Interface (TAPI)

Application Programming Interface (TAPI) allows new and existing server-based applications from Avaya's Developer Connection (DevConnect) program to work with and enhance the operation of the PARTNER Version system. The current requirement is to support a screen pop type application called "Pickup IP" which is also supported on IP Office.

The TAPI API only supports 1st Party TAPI applications.

System Programming Procedures

Code	Feature		Code	Feature
#101	System Date		#321	VMS Cover Rings
#103	System Time		#322	Remote Call Forwarding

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Code	Feature		Code	Feature
#104	Number of Lines		#323	Personalized Station Ringing
#105	Transfer Return Rings		#399	Copy Settings
#107	Recall Timer Duration		#401	Outgoing Call Restriction
#109	Outside Conference Denial		#402	Toll Call Prefix
#119	Ring on Transfer		#403	System Password
#123	Backup Programming—Automatic		#405	Disallowed List Assignments
#124	Backup Programming—Manual		#406	Emergency Phone Number List
#125	Restore Programming		#407	Allowed Phone Number Lists
#126	Automatic Daylight/Standard Times		#408	Allowed List Assignments
#127	Hold Reminder Tone		#409	Forced Account Code List
#128	Network Time Synchronization		#501	Pickup Group Extensions
#203	Hold Disconnect Time		#502	Calling Group Extensions
#205	Direct Extension Dial Lines		#503	Night Service Button
#206	Group Call Distribution		#504	Night Service Group Extensions
#208	Line Coverage Extension		#505	Hunt Group Extensions
#301	Line Assignment		#506	VMS Hunt Delay
#303	Display Language		#507	VMS Hunt Schedule
#304	Automatic Extension Privacy		#601	Fax Machine Extensions
#305	Abbreviated Ringing		#602	Music-On-Hold
#306	Transfer Return Extension		#603	Hotline
#307	Forced Account Code Entry		#604	Doorphone Extension 1
#308	Distinctive Ring		#605	Doorphone Extension 2
#309	Intercom Dial Tone		#606	Doorphone Alert Extensions
#310	Automatic VMS Cover		#610	SMDR Output Format
#311	External Hotline		#612	Contact Closure Group
#316	Call Waiting		#613	Contact Closure Operation Type
#317	Caller ID Log Answered Calls		#617	Loudspeaker Paging Line
#318	Caller ID Call Log Line Association		#728	System Reset—Programming Saved
#320	Call Coverage Rings		#730	Remote Administration Password

Centralized Telephone Programming

Extensions 10 and 11 can program other extensions on the system from their telephones.

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Chapter 3.
IP Office Norstar® Version

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3. IP Office Norstar® Version

The IP Office Essential Edition – Norstar® Version (hereafter referred to as Norstar Version) continues the evolution of communications systems for small businesses.

Norstar Version is designed to work directly out of the box without any programming or administration, allowing users to immediately make and place calls via analog trunks.

Norstar Version provides the end user a similar desktop experience as the PARTNER ACS: e.g. 2 dedicated intercom buttons, key system functionality, Caller ID, etc. This helps minimize training requirements and provides the end user with the same comfort level they have come to expect with the PARTNER ACS system.

Norstar Version provides the bulk of the feature set that the PARTNER ACS R8 supported; plus integrates enhanced features from the IP Office Essential Edition platform, such as Mobile Twinning, Visual Voice Mail, 64-party Conferencing, SIP Trunking, Full PRI/T1 and more.

Norstar Version provides the same 2-digit extension numbering plan as PARTNER ACS (default), and now supports a 3-digit flexible numbering plan (extensions 100 – 599).

Norstar Version supports all T-series and M-series Nortel telephones, the 1400 series digital telephones, the 7406E and digital Mobility Wireless products, as well as most analog devices (fax, wireless headsets, analog telephones). The original MLS series telephones and older wireless telephones are not supported.

Norstar Version also supports a new line of Avaya 9500 series digital telephones.

Norstar Version includes Embedded Voicemail; providing a 2-port Automated Attendant (upgradeable to 6-ports) and voice mail coverage activated for all telephones. Embedded Voicemail offers a variable number of hours of storage:

- 2-ports = 15 hours of storage
- 4-ports = 20 hours of storage
- 6-ports = 25 hours of storage

System Capacities

Norstar Version supports larger telephone and trunk capacity as follows (may not be attained simultaneously):

- 100 Telephones:
  - Up to 100 digital telephones
  - Up to 100 analog telephones
- 64 Trunks:
  - Up to 32 analog CO lines
  - Up to 12 BRI channels
  - 1 PRI/E1 (30 lines)
  - Up to 20 SIP lines

"Quick Mode / "Standard Mode"

The IP Office Essential Edition a-Law SD card includes two new modes of operation – Quick Mode and Standard Mode.

When the IP500 V2 is first started, it will boot-up in "Quick Mode", which has the same look and feel as the Norstar Version, e.g.: out of the box operations, 2 intercom buttons, key system functionality, voice mailboxes on all telephones, TUI programming support, etc. "Quick Mode" makes starting and using the system quick and easy.

Using Manager, users who need full IP Office Essential Edition functionality can simply change mode from "Quick Mode" to "Standard Mode". Once the system reboots, it re-starts in "Standard Mode" which is the way that IP Office Essential Edition works, looks and feels today.

System Programming

System programming is accomplished in 2 ways:

- Telephone User Interface (TUI) from extension 10 or extension 11; and uses the same administration codes as the Norstar system.
- Graphical User Interface (GUI) – Norstar Version provides a "Simplified Manager" mode in the IP Office Manager software; allowing you to program the system in less than 30 minutes

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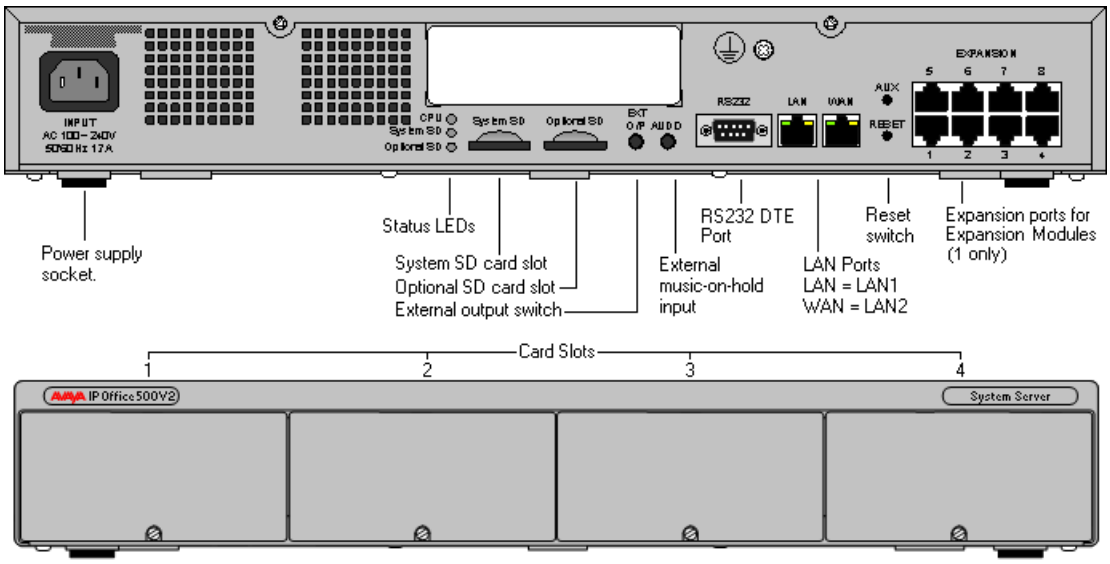
Hardware

Control Unit

Norstar Version software runs on the IP 500 V2 control unit. A Norstar Version Secure Digital SD card or IP Office Essential Ed a-law Secure Digital SD card is required and is installed in the rear of the control unit, providing operating software, Feature K Licenses and directs the system to boot up in Norstar mode or Quick Mode. An optional SD card may be installed in the option SD card slot, providing redundant backup and system upgrade capabilities.



The front of the IP500 V2 Control Unit has 4 expansion bays, which supports the following system modules:

- Combination Card – Analog Trunks
  The Combination Card provides 6 digital station (DS) ports (Ports 1-6); 2 analog station ports (Ports 7-8); 4 CO line ports (Ports 9-12) and 10 voice compression channels.
- Combination Card – BRI Trunks
  The Combination Card provides 6 digital station (DS) ports (Ports 1-6); 2 analog station ports (Ports 7-8); 2 BRI trunks (Ports 9 & 10, each trunk providing 2B+D digital channels) and 10 voice compression channels.

  Note – the system supports a maximum of 2 Combination Cards of either type.

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- DS8 Card (Max 3 per system)
  The DS8 Card provides 8 digital station ports (Ports 1-8); and supports (1) of the following optional trunk cards: ATM4, BRI4 (supports 2 BRI trunks via ports 9 & 10), BRI-8 (supports 4 BRI trunks via ports 9-12), or a PRI/E1. This card supports only the 1400 and 9500 series telephones.

  Note - each BRI trunk provides 2B+D digital channels.

- TCM8 Card (Max 4 per system)
  The TCM-8 Card provides 8 digital station ports (Ports 1-8); and supports (1) of the following optional trunk cards: ATM4, BRI4 (supports 2 BRI trunks via ports 9 & 10), BRI-8 (supports 4 BRI trunks via ports 9-12), or a PRI/E1. This card supports only Norstar T-series and M-series telephones.

  Note - each BRI trunk provides 2B+D digital channels.

- Phone 2 (Max 4 per system)
  The Phone 2 module provides 2 additional analog station ports and supports (1) of the following optional trunk cards: ATM4, BRI4, BRI8 or PRI/E1.
- Phone 8 (Max 4 per system)
  The Phone 8 module provides 8 additional analog station ports and supports (1) of the following optional trunk cards: ATM4, BRI4, BRI8 or PRI/E1.

Expansion Modules

Norstar Version supports following expansion modules:

- DS16 – provides additional 16 digital station ports (1400 and 9500 series telephones only)
- DS16A – provides additional 16 digital station ports (Norstar T-series and M-series telephones only)
- ATM 16 (Limit 1 per system) – provides additional 16 analog CO lines
- Phone 16 – provides additional 16 analog station ports
- Phone 30 – provides additional 30 analog station ports
- DS30 – provides additional 30 digital station ports (1400 and 9500 series telephones only)
- DS30A – provides additional 30 digital station ports (Norstar T-series and M-series telephones only)

Trunk Cards

Norstar Version supports the following trunk cards, which are mounted on any DS8, TCM8, Phone 2 or Phone 8 base card:

- ATM4 – provides 4 analog CO line ports. Supports Caller ID.
  Limit 4 ATM4 cards per system.
- PRI/E1 – supports full PRI or T1.
  Limit 1 PRI/E1 card per system
- BRI4 – provides 2 BRI trunks. Each BRI trunk provides 2B+D digital channels.
  Limit 4 BRI4 cards per system.
- BRI8 – provides 4 BRI trunks. Each BRI trunk provides 2B+D digital channels.
  Limit 1 BRI8 card per system.

Telephones

Norstar Version supports the following telephones:

- 1400 series digital telephones 30
- 9500 series digital telephones 31
- Norstar T-series and M-series digital telephones 31
- Wireless telephones 34

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- Analog "POTS" telephones 35

1400 series telephones

The following 1400 series telephones are supported.

- 1416 Display Telephones
- 1408 Display Telephones
- 1403 Display Ttelephone
- DBM32 Button Module

Programmable Buttons

As well as the usual dialing keys, Avaya 1400 digital telephones have dedicated function buttons like Mute, Volume, Hold, Conference and Transfer. In addition to these, there are keys that can be programmed with a range of selected special functions. These keys can be used for calling other extensions on the system (Direct Station Select or DSS keys), or can be used for options from speed dialing numbers to controlling features such as Do Not Disturb. Many features use an indicator to show whether a feature is enabled.

Button programming is done through the TUI or Simplified Manager as part of the system configuration.

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9500 series telephones

The following 9500 series telephones are supported.

- 9504 Display Telephone
- 9508 Display Telephone
- BM12 Button Module

9504

9508

BM12

Programmable Buttons

The Avaya 9500 series telephones are “paper label free”, providing programmable labels in the display. The 9504 has 4 buttons (with Red & Green LED’s) and has 2 “pages” providing a total of 12 programmable buttons. The 9508 has 8 buttons (with Red and Green LED’s) and has 2 “pages” providing a total of 24 programmable buttons.

As well as the usual dialing keys, Avaya 9500 digital telephones have dedicated function buttons like Mute, Volume, Speaker, Headset, Call Logs and Contacts. In addition to these, there are keys that can be programmed with a range of selected special functions. These keys can be used for calling other extensions on the system (Direct Station Select or DSS keys), or can be used for options from speed dialing numbers to controlling features such as Do Not Disturb. Many features use an indicator to show whether a feature is enabled.

Button programming is done through the TUI or Simplified Manager as part of the system configuration.

Norstar T-Series and M-Series telephones

The following Norstar telephones are supported on PARTNER Version:

- T-Series Telephones: T7316E, T7208, T7100, T7000 and the T24KIM (not shown)

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- M-Series Telephones: M7324, M7310 and the CAP/KLM Button Module

Wireless Telephones

T7406E Wireless

Digital Mobility 7400/4100 Series Wireless

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Note : For further details on the telephones, please refer to the telephones chapter.

Analog "POTS" telephones

The system supports a wide variety of industry standard analog telephones and devices, for example: fax machines, credit card readers; wireless headsets, cordless telephones.

Analog devices are connected to the system:

- To one of the analog ports on the Combination Card
- Via a Phone 2 or Phone 8 Module

The major features work the same, regardless how they are connected: e.g. Hold, Conference, Transfer, Call Waiting, Message Waiting, Incoming Caller ID.

Analog telephones have access to all dial-code features familiar to PARTNER users: e.g. Call pickup, Line pickup, Call Forwarding activation/deactivation, etc.

Analog telephones connected to a phone module or Combination Card default to intercom dial tone. To place an outgoing call, the user will dial 9. Analog telephones connected to an ETR module follows programmed Automatic Line Selection.

System Administration

The Norstar Version and IP Office Essential Edition in Quick Mode can be programmed in 2 ways:

- Telephone User Interface (TUI) from extension 10 or extension 11; and uses the same administration codes as the Norstar system (see Table 1).

  TUI programming is an alternate way to do System Administration and Programming on the Norstar Version. It is based on the Norstar TUI, but with improvements.

  Using TUI System Administration on ETR, 1400, 9500 and select Norstar telephones, the buttons under the display become soft-keys which are used to navigate and change the administration items. There are also shortcut codes which enable the administrator to jump to particular administration items.

  Centralized Programming is similar to System Administration, but it is used to administer buttons and other options for any telephone in the system.

  Individual Programming continues to be supported, allowing the telephone user to program buttons on their telephones.

- Graphical User Interface (GUI) – Norstar Version provides a “Simplified Manager” mode in the IP Office Manager software; allowing you to program the system in less than 30 minutes.

  “Simplified Manager” is a PC GUI application inherent to the IP Office Manager application. Although system administration can be completed via either the TUI or GUI method; some advanced features such as SIP and PRI/E1 administration require the Manager application.

  When you execute Manager, the program searches for any IP500 V2 units running Norstar Version on your network and will automatically connect using the default login and password. You are then presented with a “Simplified Manager” Screen:

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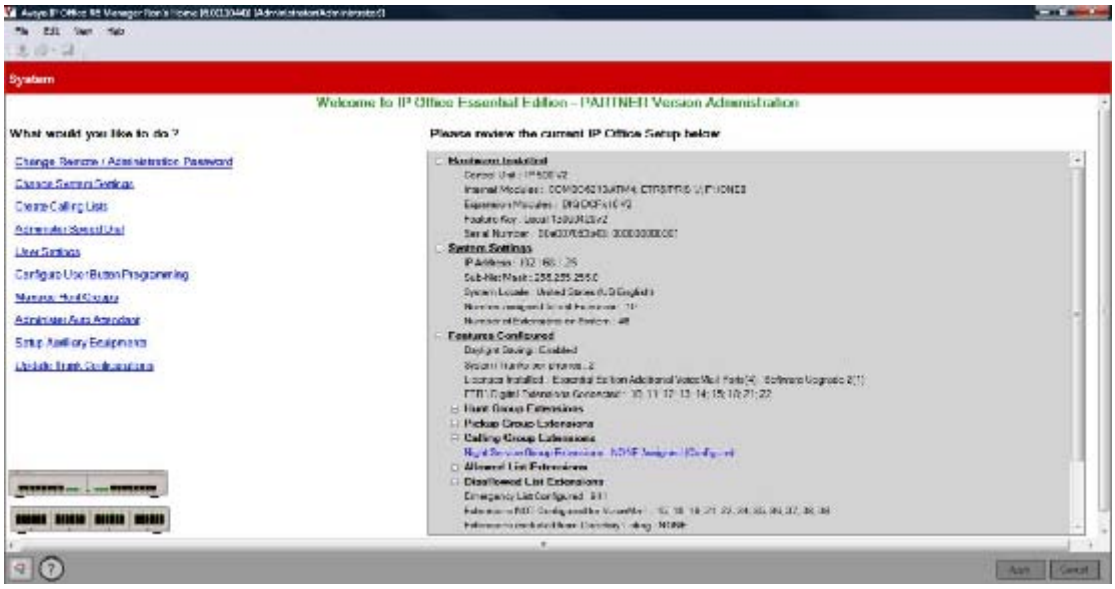
System
Welcome to IP Office Essential Edition – PARTNER Version Administration
What would you like to do ?
Please review the current IP Office Setup below

By using the system hyperlinks, you are able to quickly and easily administer the Norstar Version system in a short time. Further details can be found in the IP Office Norstar Version Manager documentation guide posted on the Avaya Support website.

System Features

The system provides the majority of the feature set that the Norstar PARTNER ACS R8 supports. A listing of the built-in features is detailed below. Feature descriptions can be found on the following pages. TUI Feature Programming Codes (if applicable) are also provided. Yellow highlighted features are new in Release 7.0.

911 Emergency Dialling	Direct Line Pickup Features	Paging Features
Abbreviated Ringing	Display	PBX Mode
Absent Text	Display Language	Personal Line Termination
Absent Text Message Inspect	Display SD Card Feature Key ID	Personalized Station Ringing
Account Codes	Display System IP Address	POT Support
Allowed/Disallowed Lists	Display System Release	Privacy
Application Programming Interface (TAPI)	Distinctive / Differentiated Ring	Recall
Attendant/Operator Positions	Do Not Disturb (Send All Calls)	Redialing Features
Auto Attendants - Emergency Greeting	Do Not Disturb Override	Remote Call Forwarding / Mobile Twinning
Auto Attendants - Multiple	Doorphones (max of 2)	Remote Access to Modem
Automatic Callback	Emergency Phone Number List	Ringing Line Preference
Automatic Daylight Savings Time	External Hotline	Ring on Transfer

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Automatic Extension Privacy	Extension Hunt Groups	SD Card Startup / Shut Down via TUI
Auto Dialing	Extension Name Display	SMDR
Automatic Line Selection	Fax Machine Extensions	Speed Dial Features
Automatic VMS Coverage	Feature Button	Station Lock/Unlock
Backup Failure Alarm	Flexible Dial Plan	Sub-Menus (see Auto Attendants)
Backup Programming Automatic	Group Calling – Ring/Page	System Password
Backup Programming Manual	Group Hunting – Ring/Voice Signal	System Groups
Bridging (Joining Calls)	Group Pickup	Transfer / Transfer Return
Call Coverage	Hold	Transfer Return to Programmable Extension
Call Forwarding - Internal	Hold Release on Abandon	Unique Line Ringing with Override
Call Log (Digital Sets only)	Hold Reminder Tone	VMS Cover Button
Call Logs (ETR sets)	Hot Dial	VMS Cover Ring Interval
Call Pickup	Hotline	VMS Hunt Delay
Call Timer	Idle Intercom Timeout	VMS Hunt Schedule
Call Timer Continuity	Intercom Manual Dial	Voice Mail
Call Waiting	Intercom Dial Tone	Bypass Greeting
Caller ID Features	Line Coverage Extension	Change Message Status to New
Cell Phone Connect (see Mobile Twinning)	Line Ringing Options	Choice of Language
Centralized Telephone Programming (from admin phones)	Message Alert Notification	End of Recording Options
CO Disconnect Time (Programmable)	Music On Hold	Outcalling Notification
Conference Drop	Mobile Twinning	Phantom Mailboxes
Contact Closure Support (2 devices)	Network Time Synchronization	Voice Mail to E-mail
Copy Settings	Night Service	Voice Mailbox Transfer – Direct
Direct Inward Dialing (DID)	On Hook Dialing	Voice Messaging
DID to Auto Attendant	One Touch Transfer	Visual Messaging

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	Wake-up Service

911 Emergency Dialing

911 Emergency Dialing can be made from any extension, regardless of any restrictions.

Abbreviated Ringing

This feature activates or deactivates Abbreviated Ringing at the system telephone at a specific extension. When you are on a call and Abbreviated Ringing is Active, any incoming call rings only once. The green light next to the line button flashes until the call is answered or the caller hangs up (or for a transferred call, until the call returns to the transfer return extension).

This feature prevents incoming calls from distracting you when you are busy on another call. To allow calls to ring repeatedly, set Abbreviated Ringing to Not Active.

Absent Text Messaging

This feature allows you to post a message (such as “Do Not Disturb” or “Away from desk”) on the display of your PARTNER ETR or digital system telephones. When another extension calls your extension, your active Absent Text Message appears on the caller’s display.

The system provides 15 pre-defined messages, plus 2 that may be customized by each user. Pre-defined messages include:

Back Soon	Back Tomorrow	Do Not Disturb
On Vacation	Out to Lunch	At Home
Away from Desk	Be Right Back	Please Call
On Holiday until	Meeting until	Don’t disturb until
With visitors until	With cust. til	At lunch until

Absent Text Message Inspect

The Inspect feature is a soft-key option, and allows you to remotely inspect the active Absent Text Message at another user’s extension.

Pressing the Absent Text Message button will provide you with an INSP option in the display. Press the button associated with the Inspect option and you are prompted to dial the extension number or press the Auto Intercom button you wish to view. When you do, the active message for that user is displayed.

Account Code Entry

Account Code Entry (ACE) is used to enter an account code (up to 15 digits) for an incoming or outgoing call currently being handled.

The Account code is printed in the SMDR record for the call for the customer to use for cost tracking purposes.

There are two different ways the feature can be used:

- Voluntary — Account Code Entry is optional.
- Forced – Account Code Entry is mandatory before dialing on an idle CO facility.

Forced account codes are validated against a list of pre-defined account codes which is created via the Manager application.

Allowed Lists / Disallowed Lists

Allowed Lists Use this feature to specify telephone numbers that users can dial regardless of other dialing restrictions, as long as they have access to an outside line. For example, if you restrict an entire category of calls through Disallowed Phone Number Lists (#404), you can permit calls to a specific number in that category by placing that number on an Allowed Phone Number List.

The System Administrator can create up to eight Allowed Phone Number Lists of up to 10 telephone numbers each.

Disallowed Lists allow you to specify telephone numbers that users cannot dial. For example, you may want to prevent calls to a specific telephone number or to categories of numbers, such as international numbers.

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The System Administrator can create up to eight Disallowed Phone Number Lists of up to 10 telephone numbers each.

Allowed and Disallowed Phone Numbers can be up to 12 digits long and may include 0–9, #, *, and 'hold' (a wildcard character, displayed as "!").

Attendant Positions

Extension 10 and Extension 11 serve as system attendant positions. TUI System Administration can only be done at these extensions. Night Service and Unlock capabilities can only be initialized from Extension 10. When a user dials 0, Extension 10 will ring.

Auto Attendants (Multiple) / Sub-Menus / Emergency Greeting

The system supports up to 9 Automated Attendants / Sub-menus. A sub-menu is the same group of selector codes as is used in an Auto Attendant.

Each Auto Attendant has its own greeting (Morning / Afternoon / Evening), its own language (English / French / Spanish), its own set of selector codes (based on its time profile), and separate VMS Hunt and Delay Schedules.

Each Auto Attendant supports its own Emergency Greeting announcement, which can be remotely recorded and activated, and plays before the Auto Attendant Greeting.

Each Auto Attendant can be accessed by dialing the appropriate Auto Attendant dial codes (to record greetings, menu prompts, etc.).

Auto Dialing

Enables a user to dial outside numbers, extension numbers, feature codes or account codes with a single touch by pressing a programmed button.

3 types of auto dial are supported:

- Auto Dial – Outside
- Auto Dial – ICM (internal extension)
- Auto Dial – ICM Page (voice signaling)

Automatic Callback

When calling an extension that is busy, press the Auto Callback softkey and the system will call you when the extension becomes free.

Automatic Daylight Savings Time

This feature automatically updates the system clock for annual Daylight Savings Time and Standard Time changes. Turn this feature off if you live in an area without daylight savings time.

Automatic Extension Privacy

Prevents other extensions with access to the same line from joining a call from their extension. Use this feature for extensions connected to a modem, fax, or any device whose function can be disrupted by someone trying to join it.

Automatic Line Selection

Specifies the order in which the system selects an available line (intercom or outside) when a user at the extension lifts the handset or presses SPEAKER to make a call without first selecting a specific line button.

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For extensions with analog telephones, set Automatic Line Selection to intercom first. This enables analog telephones to access system features, including intercom calling. When users lift the handsets on analog telephones, they hear intercom dial tone. To access an outside line, they must dial 9.

Automatic Voice Mail Coverage

Use this feature to program a button to turn VMS Cover on and off, routing unanswered intercom and transferred calls for a users' extension to the Call Answer Service of the voice messaging system after the specified number of rings (factory setting is three rings).

Backup Failure Alarm

When an Automatic Backup fails, a 'Backup Failed' message is displayed on Extension 10 and 11.

Backup Programming - Automatic

The system does a nightly backup to the system SD card. No manual intervention or external PC is required.

Backup Programming - Manual

The administrator may do a manual system programming backup to the system SD card via the TUI.

Bridging (Joining Calls)

A user can bridge (join) on any active call by pressing the associated CO line button (assuming Privacy is not active on the originating extension.) When a call is bridged, the red and green LED's will alternate on all telephones which are bridged on the call.

Call Coverage

This feature is used to redirect all Intercom, Transferred, DID and outside calls from a user's extension to another extension.

When Call Coverage is activated, covered calls are routed to the covering extension after a specified number of rings.

Call Forwarding - Internal

The Call Forwarding provides a means of redirecting Intercom (ICOM), CO, and Transfer calls from one extension to another specified extension.

Call Log - Digital Telephones

Call Log is a fixed button on digital telephones and provides a visual record of calls made and received, including unanswered calls. Details are store for both users (maximum 30 entries) and hunt groups (maximum 10 entries). The method of operation varies according to the phone type but in all cases the call records can be used for return calls.

Call Logs - ETR Telephones

Use this feature to program extensions to log answered Caller ID calls so that calls answered at an individual extension can be viewed in the Caller ID Call Log. Once the feature is activated for an extension, when a user activates the Caller ID Call Logging and Dialing (F23), all Caller ID calls that are answered by that extension are logged.

Up to 3 extensions can be designated to log all calls, both answered and unanswered, regardless of where they terminated in the system.

Call Timer Display

Displays the duration of a call from the time it was answered.

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Call Waiting

Call waiting applies to analog, single line telephones, and applies to the following types of calls:

- Intercom calls
- Transferred calls
- Forwarded calls
- Covered calls
- Outside calls (if the extension has a line appearance of the line)
- Voice Signaled calls

Caller ID Features

The following Caller ID features are supported:

- Caller ID Inspect (FEATURE 17)
  Allows you to view Caller ID information for a second call, without disconnecting the current call or putting it on hold. Use this feature to inspect ringing, active, or held calls.
- Caller ID Name Display (FEATURE 16)
  When an incoming call is ringing at a display telephone, both the name and number appear on the user's display. The Number is displayed on the top line, and the Name on the 2nd line. In some cases users may wish to see the Caller ID Name on the top line and the Caller ID Number on the 2nd line. This feature provides this capability.

Cell Phone Connect

See Mobile Twinning 48.

CO Disconnect Time (Programmable)

Use this feature to change the hold disconnect time for an outside line. Different telephone companies use different length signals. The length of the signal is called the CO or Hold Disconnect time. If you place a call on hold and the caller hangs up, but the call does not disconnect within a minute after the caller.

Conference Drop

On 1400 and 9500 digital telephones, you may choose which specific party you want to drop from a conference call. On ETR and analog telephones, the last party added to the conference call is automatically dropped.

Conferencing

The system supports up to 64-party conferencing. A maximum of 2 analog lines can be in any conference call. Conference calls may consist of combinations of analog lines, digital lines (PRI/T1, SIP), analog telephones, ETR telephones or digital telephones.

Contact Closure Support for 2 Devices

The IP500 V2 Control Unit has a Contact Closure control jack on the back panel. This is a 3.5 mm stereo jack that can drive two external adjunct relays.

This feature allows users at eligible extensions to control an external adjunct by entering feature codes at their system or standard telephone. The port can control a variety of devices, the most common being door strikes, night bells, lighting, or voice recording systems.

Copy Settings

This feature allows a System Administrator to copy extension-specific programming from one extension to another. When the Extension Copy operation is performed, the destination extension will have the same properties as the originating extension.

Extension Copy can only be performed from an Attendant Extension in System Programming Mode or from the Windows Manager PC application.

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Dial Code Features

- 'F' represents the FEATURE button.
- 'I' represents the INTERCOM button.
- 'XX' represents a system extension number.
- 'PP' represents a Personal Speed Dial code.
- 'G' represents a single-digit group number.
- 'LL' represents a system line number.

Feature	Partner Version Code	Norstar/BCM Code
Do Not Disturb	F01	F85
Do Not Disturb Cancel	F01	F#85
Recall	F03	F71
Save Number Redial	F04	F67
Last Number Redial	F05	F5
Conference Drop	F06	F934 (new)
Privacy	F07	F83
Call Forward	F11	F4
Call Forward Cancel	F11	F#4
Account Code Entry	F12	F900
Voice Mailbox Transfer	F14	F986
VMS Cover	F15	F984
Caller ID Name Display	F16	F933 (new)
Caller ID Inspect	F17	F811
Call Coverage	F20	F932 (new)
Station Lock	F21	F936 (new)
Station Unlock	F22	F937 (new)
Caller ID Log	F23	F812
Hot Dial	F26	F935 (new)
Message Alert Notification	F27	F938 (new)
Absent Message	F28	F931 (new)
Contact Closure 1	F41	F9*41
Contact Closure 2	F42	F9*42
System Release	F590	F9*80
System IP Address	F591	F9*81

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SD Card Serial Number	F592	F9*82
System Speed Dial	F6XX (00-99)	F06XX (00-99)
Personal Speed Dial	FXX (80-99)	F*4XX (80-99)
DND Override Numbers	F7XX (00-19)	F07XX (00-19)
Call Transfer	Fixed Button	F70
Voice Announce to Extension	I* + XX	F66 + XX (new)
Call Park to own Extension	Transfer + XX	F74 (new)

Direct Inward Dialing (DID)

Use this feature to have Direct Inward Dialing (DID) numbers on a PRI / T1 or SIP line routed automatically to a specific extension, hunt group, or calling group. When a Direct Inward Dialing (DID) call arrives, the PARTNER system “collects” the digits from the T1 service provider (that is, the number that was dialed) and matches the number to a DID Mapping Table you created. If the incoming number matches an entry in the DID Mapping table, the call rings at the matching extension or hunt group.

Direct Inward Dialling (DID) to Auto Attendant

DID calls can be directed to and answered by any Automated Attendant.

Direct Line Pickup

With the Direct Line Pickup features, you can access a ringing or held call or a call in progress. You can also select a line to use that is not assigned to your telephone. The Direct Line Pickup features are:

- Direct Line Pickup—Active Line
- Direct Line Pickup—Idle Line

Display Language

All system telephones offer the choice of 3 display language:

- US English
- Canadian French
- Latin American Spanish

Each individual extension may select their own language.

Display SD Card Feature Key ID

Displays the SD Card Feature Key ID on any display telephone.

Display System IP Address

Displays the IP address currently assigned to the IP500 V2 system.

Display System Release

Displays the software release running on the IP500 V2 system.

Distinctive / Differentiated Ringing

The system provides different ring patterns for external calls, internal calls and transferred calls.

Do Not Disturb

Use this feature to be able to press a programmed button to prevent incoming calls for the extension from ringing (lights still flash). When Do Not Disturb is on, outside callers hear ringing while inside callers hear a busy signal. Calls are immediately sent to your voice mailbox.

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Do Not Disturb Exceptions

Allows a user to input a list of up to 20 telephone numbers and/or internal extension numbers that will ring at the users' station when Do Not Disturb is activated.

Doorphones

A doorphone is usually placed near an entrance, to screen visitors. You can connect up to two doorphones to the system.

Emergency Phone Number List

Create a list of up to 10 telephone numbers that all users can dial regardless of dialing restrictions, provided they have access to an outside line. Typical list entries include fire, police, and emergency services numbers. police, and emergency services numbers.

Extension Hunt Groups

Use this feature to assign any number of extensions to a Hunt Group. When extensions are in a Hunt Group, an incoming call searches or "hunts" for the longest idle extension that is available. The system supports up to six Hunt Groups

Extension Name Display

The user' name and extension number is displayed on the phone display.

External Hotline

When a user lifts the handset of an external hotline, a predetermined outside number is dialed automatically. The external number might be, for example, a frequently called service bureau. The external hotline must be a single line telephone, not a system telephone, and should not have a dial pad.

Fax Machine Extensions

Allows you to connect your fax machine to your system, allowing you to share network facilities.

Flexible Dial Plan

Through Manager, the system defaults to a fixed 2-digit dial plan (extensions 10 – 57); but can be changed to a flexible 3-digit dial plan (extensions 100 – 599).

Group Calling - Ring / Page

Use this feature to ring, page, or transfer calls simultaneously to all of the extensions in any one of four Calling Groups. (G represents a Calling Group number from 1–4.) When paging, you hear a beep and can begin speaking. Your voice is heard on the speakers of all idle system telephones in the selected Calling Group. The first extension to answer the call (on ETR sets - by picking up the handset or pressing SPEAKER, on 14xx sets - or by pressing the ANSWER softkey on the display) is connected to the caller.

You can group page employees for general announcements; this is an inexpensive alternative to a paging system.

Group Hunting - Ring / Page

Use this feature to distribute call volume among extensions in a group, to off-load call activity from a single user. When an intercom or transferred call is placed to a Hunt Group (where G is a Hunt Group number from 1–6), the system rings or voice-signals the first available extension in the group, passing over busy extensions (or those with Do Not Disturb on) in a circular hunt. If a ringing call is not answered within three rings, the call moves to the next available extension, and so on, until the call is answered or the caller hangs up.

For a voice-signaled call, only the first available extension is signaled; if that extension does not answer, the call does not keep hunting. After an extension receives a Hunt Group call, the next call to the Hunt Group will not ring or voice-signal that extension first unless all other extensions in the group are busy or do not answer.

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Group Pickup

Use this feature to answer any outside, intercom, or transferred call ringing at an extension in a Pickup Group (where G is a Pickup Group number from 1–4). When a call rings at an extension that is in a Pickup Group, you can answer the call at any extension without knowing which extension or line is ringing and without being in the same Pickup Group. The system can have up to four Pickup Groups.

Hold Reminder

If a call is left on hold for longer than a specified period, the extension will ring back, alerting you that the call is still on hold. This ring continues to sound until the held call is retrieved, or until the caller hangs up.

Hot Dial

This feature enables a user to start dialing a number without first pressing the speaker button or going off-hook.

Hotline

This feature allows designated extensions to automatically place an intercom call to another designated extension, when it goes off-hook.

Intercom Dial Codes

- 'I' represents the INTERCOM button.
- 'XX' represents a system extension number.
- 'G' represents a single-digit group number.
- 'LL' represents a system line number.

Intercom Dial Code	Feature
I6XX	Call Pickup
I66G	Group Pickup
I68LL	Direct Line Pickup—Active Line
I70	Loudspeaker Paging
I*70	Simultaneous Paging
I7G	Group Calling—Ring
I*7G	Group Calling—Page
I77G	Group Hunting—Ring
I*77G	Group Hunting—Voice Signal
I8LL	Direct Line Pickup—Idle Line

Intercom Dial Tone

Use this feature to determine the type of dial tone that the system provides at an extension. It may be necessary to change this setting to Machine (outside line) dial tone for an autodialing device, such as a fax or modem, that has trouble making calls. For example, if you have a modem that checks for outside line dial tone before dialing, use this procedure to change from Regular to Machine dial tone.

Intercom Manual Dialing

Users can make an internal call to another extension by ringing it (dialing its extension number) or voice signaling the extension by prefixing the extension number with a *.

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Line Coverage Extension

Use this feature to identify an extension as the “owner” of a specific outside line, so the extension can activate Call Coverage or VMS Cover for that line.

Line Ringing Options

This feature specifies the options: Immediate Ring, Delayed Ring or No Ring for incoming calls on a line appearance programmed on the extension.

Log All Calls

Up to 3 extensions can be designated to log all calls, both answered and unanswered, regardless of where they terminated in the system.

Message Alert Notification

Message Alert Notification allows any ETR or Digital telephone, programmed with Auto Intercom buttons, to see which users have new messages waiting in their mailbox.

When the Message Alert Notification button is pressed, the DSS/BLF buttons programmed on the user’s telephone will display a flashing red LED when the associated user has new voice mails waiting.

This feature may be left on at all times or deactivated by pressing the Message Alert button again.

Music on Hold

Connect an audio source to the Music-On-Hold port on the processor module so that a caller hears music when placed on hold when the Music-On-Hold feature has been activated.

Network Time Synchronization

Use this feature to specify whether you want to synchronize the system clock with the network time delivered by your service provider over Caller ID. If Network Time Synchronization is set to Active, the system will use the network time provided by your service provider; if it is set to Not Active, the system will use the time provided by its system clock.

Night Service

When Night Service is on and a call comes in, all extensions in the Night Service Group ring immediately, regardless of normal Line Ringing settings.

One-Touch Transfer

This feature allows a user to automatically transfer a connected call with one-touch -- by pressing a pre-programmed Auto Dial Intercom for the destination extension. (The *Transfer* button is not used.)

- The transfer is completed when the transferring station goes on-hook, or touches the *Transfer* button or *Complete* soft-key.
- The user initiating the transfer is able to consult and complete the transfer or blind transfer after successfully initiating a one-touch transfer.

The LEDs at the phone initiating the transfer, as well as other appearances of the call in the system, behave the same as if the user touched the *Transfer* button to initiate the transfer.

Paging Features

- Loudspeaker Paging - Users can initiate a call (either manually dialed or intercom auto dialed) to an extension connected to an analog station port that connects them to an external loudspeaker paging system.
- Simultaneous Paging is a feature that pages both the external loudspeaker paging system (via the paging port) as well as voice signals all of the extensions in Calling Group 1.

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PBX Mode

The system can operate in Key System mode (default) where all lines appear on all telephones; or can be administered to operate in PBX-mode. PBX-mode provides 3 Intercom buttons per telephone which work similar to System Access buttons (provides intercom dial tone, d1al 9 to access an outside line).

Note: On PARTNER ETR sets, the system provides 2 Intercom buttons per telephone.

Personal Line Termination

Allows you to assign a line / lines to specific extension / extensions.

Personalized Station Ringing

Allows you to assign a personalized ring pattern to an extension

POTS (Plain Old Telephone Set) Support

POTS (single line analog telephones) are supported on the system. They are connected to an analog station port on the Combo module; a port on the Phone-2, Phone-8 or Phone-16 module, or to an ETR port.

Privacy

Privacy prevents other users from joining on your call.

Recall

Use this feature to send a timed switch hook flash over the telephone line to “recall” a new dial tone or to access certain central office features, such as Call Waiting or 3-way calling.

Redialing Features

There are 2 features which offer redial functionality:

There are 2 features which offer redial functionality:

- Last Number Redial (FEATURE 05): Last Number Redial redials all digits dialed on the last outside call except account codes.
    - On 1400 series telephones, there is a fixed Redial button, which allows you to scroll through and redial the last 20 number called from that telephone set..
- Save Number Redial (FEATURE 04): Saves into temporary memory the last outside number (up to 28 digits) dialed from a system telephone. Use this feature to save a number before you hang up on a busy or unanswered call. Once saved, the number can be redialed at any time. The number stays in memory until a different one is saved.

Remote Call Forwarding / Mobile Twinninng

There are two types of Remote Call Forwarding supported on PARTNER Version:

1. Remote Call Forwarding (FEATURE 11) allows a user to send internal and external calls destined for a particular extension to an external phone number (off the system) over an analog trunk. Remote Call Forwarding will not alert the internal extension, but immediately sends the call out to the predefined number.
2. Mobile Twinning allows a user to send internal and external calls destined for a particular extension to an external phone. The call simultaneously rings at the internal extension and receives the extension’s coverage (including Voicemail) treatment if the call has not been answered. This feature is only available on systems that have digital trunks (SIP, PRI/T1).

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Remote Access to the Built-in Modem

This feature allows for unattended access to the built-in modem (x76) by calling into the system:

- By a DID number, or
- By transferring to the modem via the Automated Attendant.

When an Automated Attendant Selector code with the "Transfer To Number" action has the modem as its destination, the system will transfer the call to the built-in modem.

Ring on Transfer

This is used to specify what the caller hears while they are being transferred. If Ring on Transfer is set to Active, callers hear ringing while they are being transferred; if it is set to Not Active callers hear beeps or Music on Hold if this is activated and a music source is connected to the system.

Ringing Line Preference

When an extension is on-hook and the extension is ringing, the user simply goes off-hook to answer the call and is automatically connected to the ringing call. If more than one call is alerting, the system automatically connects the user to the longest ringing call.

SD Card Start-up / Shut Down via TUI

A new TUI command is provided to shut down and restart the system SD card slot and the optional SD card slot, without shutting down the entire system. This allows for removal and reinstallation of the system SD and optional SD cards.

Speed Dial Options

The system supports the following types of Speed Dialing:

- Personal Directory - 1400 and 9500 Telephones only
  Each user can store up to 20 frequently-dialed names and numbers in their extension. Personal Directory numbers can be dialed quickly by pressing CONTACTS and dialing the first few letters of the person's name. Personal Directory numbers programmed for a particular extension are for use only at that extension.
- Personal Speed Dial (FEATURE 80 – 99) – ETR and Analog Telephones only.
  Each user can store up to 20 frequently-dialed numbers (no name) in their extension. Personal Speed Dial numbers can be dialed quickly by pressing FEATURE (or # at intercom dial tone on a single-line telephone) and the two-digit speed dial code from 80 through 99. The Personal Speed Dial numbers programmed for a particular extension are for use only at that extension.

  On analog telephones, a user will press the # key (instead of the Feature key) then dial the 2-digit number.
- System Speed Dial – 1400 and 9500 Telephones only
  System Speed Dialing is a shared list of up to 100 frequently-dialed names and numbers (FEATURE 600 – 699). All users on the system may use these numbers by pressing CONTACTS and dialing the first few letters of the party's name.
- System Speed Dial (FEATURE 600 – 699) – ETR and Analog Telephones only.
  System Speed Dialing is a shared list of up to 100 frequently-dialed numbers (no name) up to 28 characters in length. All users on the system may dial a System Speed Dial number by pressing FEATURE (or # at intercom dial tone on a single-line telephone) and the three digit speed dial code, from 600 through 699

Station Lock / Unlock

Station Lock helps to prevent unauthorized people from making outside calls from extensions. Users enter a four-digit code on their telephone dialpad to "lock" your extension. To unlock the phone, the user enters the identical code.

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Station Message Detail Reporting (SMDR)

SMDR is a call reporting feature that provides records of call activity. It is commonly used in many types of business, including Legal, Contact Centers, Sales and Real Estate. Call reporting information allows users to:

- Detect any unauthorized calls.
- Bill clients or projects.
- Bill back by department.
- Reduce telephone costs by identifying the need to change telecommunications services
- Print Caller ID information.

The output is generally sent to a PC running an optional Call Accounting software package.

System Groups

Norstar Version supports the following types of groups:

- Pickup Groups (4 Groups)
  When a call rings at an extension assigned to a Pickup Group, a user at any other extension in the system can answer the ringing call by dialing the Pickup Group code. The Pickup Group feature helps when a user needs to answer calls on lines or pools not assigned to his or her telephone.
- Calling Groups (4 Groups)
  A Calling Group is a group of extensions that can be called at the same time. Any user in the system can ring or page all extensions in a Calling Group at the same time or transfer a call to a Calling Group. The first extension to pick up the call is connected to the caller. A typical use of this feature is to have callers ring into a Calling Group of sales representatives, or to create a “Page All” group.
- Hunt Groups (6 Groups)
  When extensions are in a Hunt Group, an incoming call searches or “hunts” for the first available extension.
- Night Service Group (1 Group)
  When Night Service is activated and a call comes in, all extensions assigned to the Night Service Group ring immediately, regardless of normal Line Ringing settings.

System Password

Allows you to define a four-digit password that users can enter from system telephones to override dialing restrictions (if the extension has access to an outside line) or turn Night Service on and off. Not available on analog telephones.

Transfer / Transfer Return

You can transfer calls to an extension or group using the TRANSFER button on a system telephone or the switchhook on a single-line telephone.

Transferred calls return to the originating extension if they are not answered. You also can program the number of times a transferred call rings before it returns to an extension.

Transfer Return to a Programmable Extension

This feature provides an option to re-route unanswered transferred calls to an alternate extension.

- Programmable on a per extension basis.
- The existing System Programming option, *Transfer Return Rings*, will be used to indicate when transferred calls are returned to the Transfer Return extension.

If a call is routed to a transfer return extension and there are no available intercom appearances to terminate the call, the call will continue to alert at the transfer destination until the transfer return extension becomes available.

The default value for each transfer return extension is its own extension.

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Unique Line Ringing (#209) / Override Line Ringing

This feature allows users to differentiate by sound which line is ringing by assigning a ring pattern to it. Once a ring pattern is assigned to a line, incoming calls on that line ring with the assigned ring pattern.

- 8 ring patterns are available.
- Unique Line Ringing does not apply to DID calls because the channel used is not guaranteed.
- The default is pattern 1.

If the Override Line Ringing feature is active, then incoming calls on a line that ring at an extension ring with the configured extension ring pattern.

Visual Voice Messaging

Supported on 1400 and 9500 digital telephones only. Allows the user to manage their voice mail messages via their display. Features supported include:

- access new /old / saved messages.
- next and previous message.
- fast forward and rewind.
- pause message.
- save, delete and copy message to other users of the system.
- change default greeting.
- change password.
- change message status from old / saved to New.

Voice Mail Options

- Bypass Greeting
  - After a call goes to an extension's voice mailbox, the caller can press the 1 key to bypass the mailbox greeting and begin recording a message.
- Change Message Status to New (*06)
  - This feature provides the ability to change the status of a message from *Old* or *Saved* to *New*.
  - Users can dial *06 at any time after the *Old* or *Saved* message begins to play through the 3 second pause after the caller information portion of the message.
    - The message is moved to the queue of New messages.
  - When a message is marked as *New*, the system does not initiate a new Voicemail to Email message, but the extension's message waiting light is illuminated.
- Choice of Language (English / French / Spanish)
  - Each Auto Attendant can have a primary language assigned to it. The language applies to the pre-recorded greetings and prompts available to that Auto Attendant.
  - This overrides the system language setting.
  - The per-extension Display Language setting will also override the system language and change all the displays, pre-recorded greetings, and prompts to the chosen language.
- End of Recording options
  - When a caller has finished leaving a message in an extension's mailbox, the caller can touch the # key to signal the end of recording of the message.
  - If the message is less than 3 seconds, the system will play the following prompt: "Message too short. Deleted. Goodbye."
  - If the message is longer than 3 seconds, the system will play the following prompt: "Your message has been sent. Goodbye."
- Outcalling Notification
  - If you use the Outcalling option, Embedded Messaging will call the number which you have programmed to signal that a new message has arrived in your mailbox.
    - Each user can have 1 number programmed for outcalling notification.

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        - The system will make up to 3 attempts (in 5-minute intervals) to contact the user.
        - After 3 attempts, outcalling notification will cease until another new voice message arrives.
- Phantom Mailboxes
    - Provides voice mailbox support for all extensions (10-57), without requiring the physical hardware to be present. Extensions without hardware are called *phantom extensions*.
        - Note: Ports 7 and 8 of an ETR-6 module can be used for phantom extensions even though you cannot plug a phone into them.
    - Calls to a phantom extension go directly to voice mail.
        - DTMF breakout service can be used to transfer the call.
    - Phantom voice mailboxes:
        - Default like a normal mailbox.
        - Can be accessed remotely if the remote access feature of the mailbox is enabled.
        - Voicemail to Email can be activated for a phantom mailbox via the Simplified Manager.

VMS Cover

Allows you to turn VMS Cover on and off, routing unanswered intercom and transferred calls for users extensions to the Call Answer Service of the voice messaging system after the specified number of rings.

VMS Cover Ring Interval

Allows you to define the number of times a call rings before it is sent to a user's voice mailbox.

VMS Hunt Delay

Allows you to determine when outside calls should be answered by the Automated Attendant of the voice messaging system. You can set the system for any number of rings, 0–6. Assigning more rings gives the operator an opportunity to answer calls before they go to the Automated Attendant.

VMS Hunt Delay is programmable on a per-line basis. In addition, you can program this feature so that calls can be handled one way during the day and a different way when the system is in Night Service.

VMS Hunt Schedule Interval

Allows you to determine when outside lines are covered by the Automated Attendant. Options include all the time, only during Day operation (Night Service is off), or only during Night operation (Night Service is on).

Voice Mailbox Transfer - Direct

Allows you to transfer a call directly into a users voice mailbox without ringing their extension.

Voice Messaging

The PARTNER Version has Embedded Voicemail included in the system. All extensions are automatically assigned a mailbox.

Embedded Voicemail also provides:

- A 2-port, single level Automated Attendant
- Separate Morning, Afternoon, Evening and Out of Hours menu greetings with time profiles
- Dial by Name capability
- Variable hours of storage:
    - 2-ports = 15 Hours
    - 4-ports = 20 Hours
    - 6-ports = 25 Hours
- Up to 3 minutes per message

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- Fast Forward, Rewind, Replay and Skip capability
- Voice Mail to e-mail capability
- Remote message retrieval
- Visual Messaging (on digital telephones only)

Wake-Up Service

This feature allows a system phone (x10) to set a Wake-up Call on behalf of another user.

- When a Wake-up Call is scheduled, the system will place an intercom call to the target extension at the scheduled time.
  - Wake Up calls ring for approximately 30 seconds (and overrides DND).
  - If the target extension is busy on a call or has an alerting call, the Wake Up call is delivered to the phone as an intercom call.
  - If the Wake Up call is answered, the user will hear Music On Hold if it is active; otherwise, they will hear silence.
  - If the Wake Up call is not answered, the system will make a 2nd attempt 5 minutes later.
    - If the call is not answered after the 2nd attempt, then the Wake Up call is abandoned.
  - A scheduled Wake Up call occurs once in a 24 hour period.

Application Programming Interface (TAPI)

Application Programming Interface (TAPI) allows new and existing server-based applications from Avaya's Developer Connection (DevConnect) program to work with and enhance the operation of the PARTNER Version system. The current requirement is to support a screen pop type application called "Pickup IP" which is also supported on IP Office.

The TAPI API only supports 1st Party TAPI applications.

System Programming Procedures

Code	Feature		Code	Feature
#101	System Date		#321	VMS Cover Rings
#103	System Time		#322	Remote Call Forwarding
#104	Number of Lines		#323	Personalized Station Ringing
#105	Transfer Return Rings		#399	Copy Settings
#107	Recall Timer Duration		#401	Outgoing Call Restriction
#109	Outside Conference Denial		#402	Toll Call Prefix
#119	Ring on Transfer		#403	System Password
#123	Backup Programming—Automatic		#405	Disallowed List Assignments
#124	Backup Programming—Manual		#406	Emergency Phone Number List
#125	Restore Programming		#407	Allowed Phone Number Lists
#126	Automatic Daylight/Standard Times		#408	Allowed List Assignments
#127	Hold Reminder Tone		#409	Forced Account Code List
#128	Network Time Synchronization		#501	Pickup Group Extensions
#203	Hold Disconnect Time		#502	Calling Group Extensions
#205	Direct Extension Dial Lines		#503	Night Service Button
#206	Group Call Distribution		#504	Night Service Group Extensions
#208	Line Coverage Extension		#505	Hunt Group Extensions
#301	Line Assignment		#506	VMS Hunt Delay
#303	Display Language		#507	VMS Hunt Schedule
#304	Automatic Extension Privacy		#601	Fax Machine Extensions
#305	Abbreviated Ringing		#602	Music-On-Hold
#306	Transfer Return Extension		#603	Hotline
#307	Forced Account Code Entry		#604	Doorphone Extension 1
#308	Distinctive Ring		#605	Doorphone Extension 2
#309	Intercom Dial Tone		#606	Doorphone Alert Extensions
#310	Automatic VMS Cover		#610	SMDR Output Format

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Code	Feature		Code	Feature
#311	External Hotline		#612	Contact Closure Group
#316	Call Waiting		#613	Contact Closure Operation Type
#317	Caller ID Log Answered Calls		#617	Loudspeaker Paging Line
#318	Caller ID Call Log Line Association		#728	System Reset—Programming Saved
#320	Call Coverage Rings		#730	Remote Administration Password

Centralized Telephone Programming

Extensions 10 and 11 can program other extensions on the system from their telephones.

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Chapter 4.
IP Office Platform

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4. IP Office Platform

IP Office is a modular communications solution that scales from 2 to 384 extensions on a single platform. IP Office supports up to 32 sites and 1000 users in a multi-site network with resilient and redundant capabilities. It provides a hybrid PBX with both Time Division Multiplexing (TDM) and IP telephone and trunk support that can be used in either mode or both concurrently. IP Office has data capabilities built in, providing IP routing, switching and firewall protection between LAN and WAN (LAN2). IP Office has a web based applications suite that delivers contact center reporting, voice and email messaging, Interactive Voice Response, conferencing and computer telephony integration.

IP Office solutions are built from hardware units and application software. Hardware provides the connectivity for voice and data circuits and processor units for the solution software. Each IP Office solution will require a system control unit (IP500 V2), trunk connections to service provider, and expansion modules for TDM telephone cabling. IP Telephones connect over LAN connections to the IP Office solution.

With Release 7.0, new Digital Station modules will be available supporting select Nortel digital telephones. The new DS modules will support RJ21 connectors on the telephone station side. Enhanced support for select Nortel IP telephones will be available with Release 7.0. Refer to the hardware section later in this document.

IP Office Control Unit

The IP500 V2 control unit is compatible with all IP500 base extension and Voice Compression Modules (VCM) cards as well as all external IP500 expansion modules.

IP500 V2 is a stackable unit with an optional 19" rack mounting kit and an optional wall mounting kit for smaller configurations. The control unit provides:

- 4 slots to house a mixture of extension cards and VCM cards:
- Digital Station 8 card. Supporting IP Office digital telephones
- TCM8 Station card supporting select Nortel digital telephones (IP500 V2 control unit only)
- Phone 2 and Phone 8 cards
- VCM32 and VCM64 cards
- 4-Port Expansion card
- ETR6 port phone card (IP500 V2 only, North America only)
- Combination Card with 10 VCM channels, 4 analog trunks, 6 Digital Station (DS) and 2 Phone ports (IP500 V2 only)
- Combination Card with 10 VCM channels, 2 BRI trunk interfaces (4 channels), 6 Digital Station (DS) and 2 Phone ports (IP500 V2 only)
- Optional trunk card support:
- Analog Trunk Module 4 card
- BRI4 and BRI8 cards (2 x 2B+D and 4 x 2B+D channels respectively)
- Single and Dual Universal PRI cards
- IP500 Legacy Card Carrier: this card allows up to two IP400 Trunk or VCM Cards to be used with the IP500 control unit
- IP500 V2: 2 Slots for SD cards (System and optional) – System SD card is required for system operation as well as licensing of optional features and storage for embedded voicemail in Essential Edition.
- 9-pin RS232 DTE Port for maintenance.
- Support for up to 12 IP500 Expansion Modules:
  - Phone modules (16, 30)
  - Digital Station modules (16, 30) supporting IP Office telephones with RJ45 connector
  - Digital Station modules (16A, 30A) supporting Nortel telephones with RJ21 connector (IP500 V2 control unit only)
  - Analog Trunk Module 16
  - IP500 So8 module
  - IP400 expansion modules (not WAN3 10/100 or Network Alchemy modules)
- External output socket supporting two relay on/off switch ports, e.g. for door entry systems

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- Audio input port for external music on hold source
- 48 Data channels
- Up to 40 VoiceMail Pro ports
- Two 10/100 switched Ethernet ports (Layer 3)

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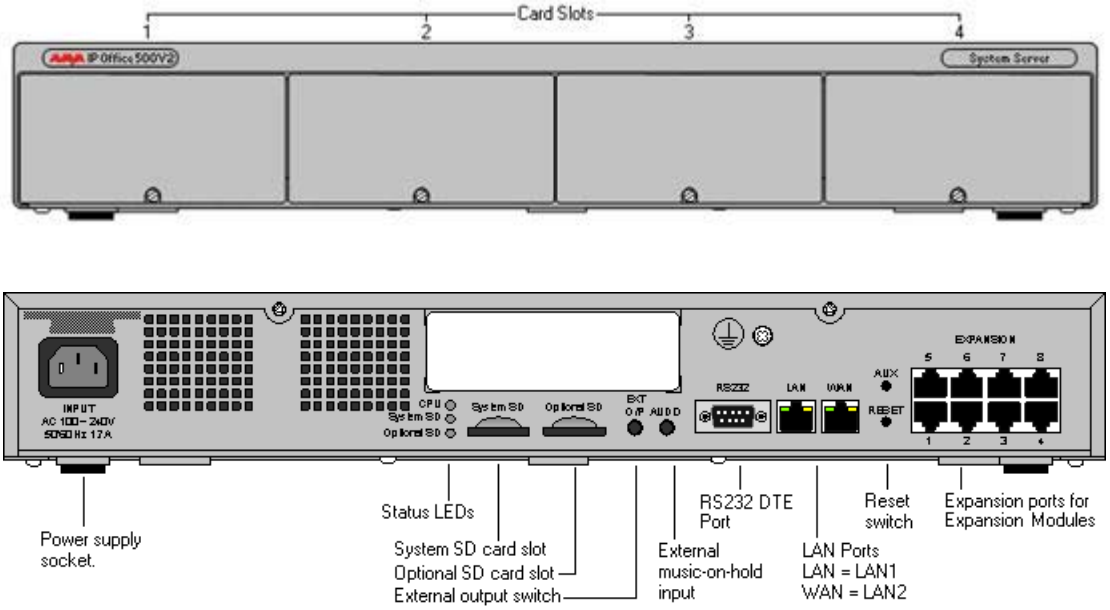

IP500 Base Cards

The IP500 control unit has 4 slots for the insertion of cards. These cards can be divided into two types; base cards and trunk cards. Base cards include a front panel and ports for cable connections. Trunk cards can be added to a base card in order to provide additional facilities (typically trunk connections).

The following base cards are available:

- IP500 Digital Station 8 Card 86 *(Maximum 3)*
- IP500 Digital TCM Station 8 Card *(Maximum 4)*
- IP500 Analog Phone 2 Card and Phone 8 Card 86 *(Maximum 4)*
- IP500 VCM Card 89 *(Maximum 2)*
- IP500 4-Port Expansion Card 90 *(Maximum 1)*
- IP500 Legacy Card Carrier 90 *(Maximum 2)*
- ETR 6 Port Phone Card 88 *(Maximum 3, IP500 V2 only)*
- Combination Card with 4 Analog Trunks 88 *(Maximum 2, no trunk card, IP500 V2 only)*
- Combination Card with 2 BRI Trunks 89 *(Maximum 2, no trunk card, IP500 V2 only)*

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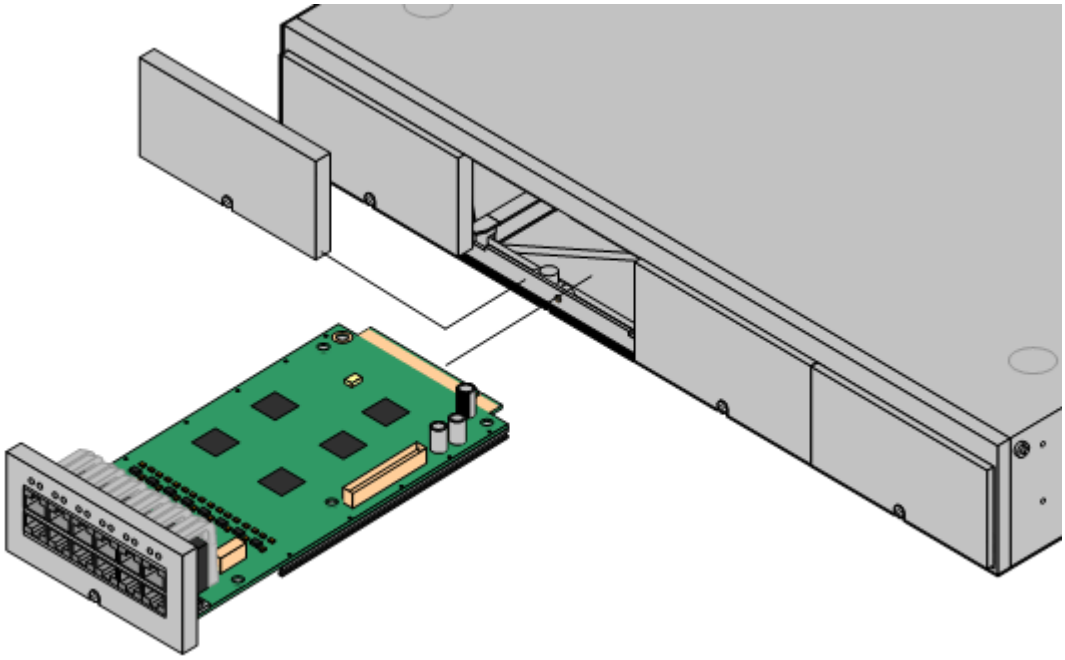

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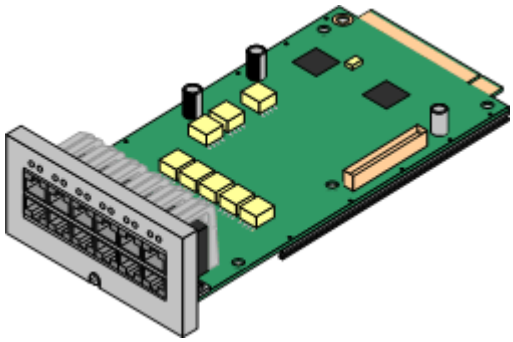
IP500 Digital Station 8 Card

This card provides 12 RJ45 ports. The first 8 ports are DS ports for the connection of Avaya digital telephones other than IP telephones. The card can be fitted with an IP500 trunk card, which then uses the additional 4 RJ45 ports for trunk connections.

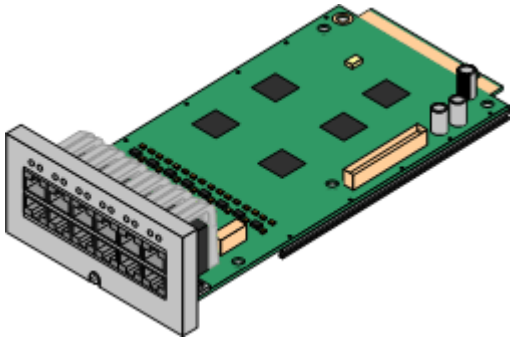
A variant of the digital station card, Digital Station TCM8, supports select Nortel digital telephones. Refer to Telephone section for details.



- This card accepts one IP500 trunk card of any type.
- This card supports selected Avaya 2400, 5400 and 6400 Series telephones, plus T3 telephones(EMEA only) and 3810 wireless telephones (North America only).
- 4400 Series telephones (4406D, 4412D and 4424D) are not supported on this card, only on Digital Station expansion modules. Therefore a maximum of 360 x 4400 Series telephones are supported in the system.

IP500 Analog Phone 2 Card

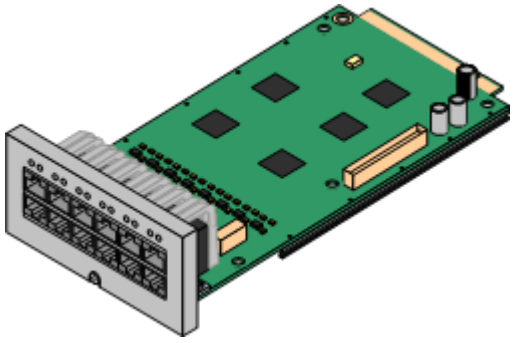
This card provides 2 analog extension ports (1-2) for the connection of analog telephones. The card can be fitted with an IP500 trunk card, which then uses the last 4 RJ45 ports (9-12) for trunk connections.



- This card accepts one IP500 trunk card of any type.

IP500 Analog Phone 8 Card

This card provides 8 analog extension ports for the connection of analog telephones. The card can be fitted with an IP500 trunk card, which then uses the additional 4 RJ45 ports for trunk connections.



- This card accepts one IP500 trunk card of any type. When fitted with an Analog Trunk card, it provides a single power-fail port.

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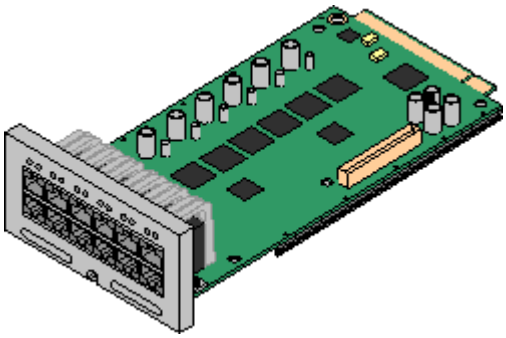
IP500 ETR 6 port Phone Card

This card provides 6 ETR ports for the connection of ETR 6/18/34 D, Euro-Style and Refresh digital telephones as well as 3910/3920 wireless telephones. The card can be fitted with an IP500 trunk card, which then uses the additional 4 RJ45 ports for trunk connections. A maximum of three of these cards is allowed in the IP500 V2 chassis.



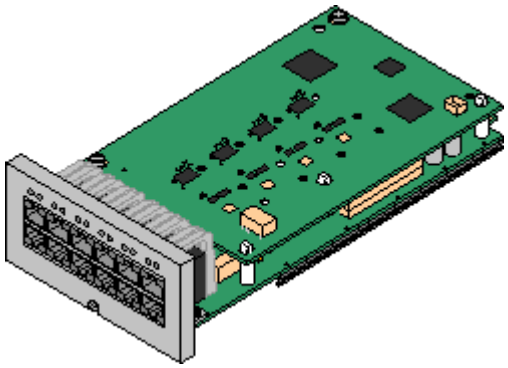
This card

- accepts one IP500 trunk card of any type. When fitted with an Analog Trunk card, it provides a single power-fail port.
- works in IP500 V2 only.
- is only supported by the IP Office Essential Edition - PARTNER Version (please see chapter 1b).

IP500 Combination Card with 4 Analog Trunks

This card works in the IP500 V2 only and provides:

- 4 Analog trunk ports.
- 2 Analog extension ports for the connection of analog telephones.
- 6 Digital Station (DS) ports for the connection of Avaya digital telephones.
- 10 VCM channels to support IP telephones or voice networking.



This card already includes an IP500 trunk card, therefore no additional trunk card is possible. It provides a single power-fail port (either port 7 or port 8). A maximum of two of these cards is allowed in the chassis.

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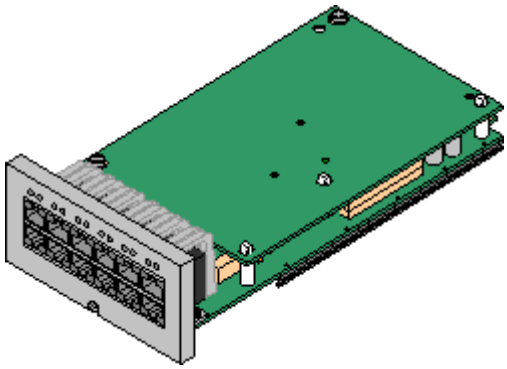
IP500 Combination Card with 2 BRI trunks

This card works in the IP500 V2 only and provides

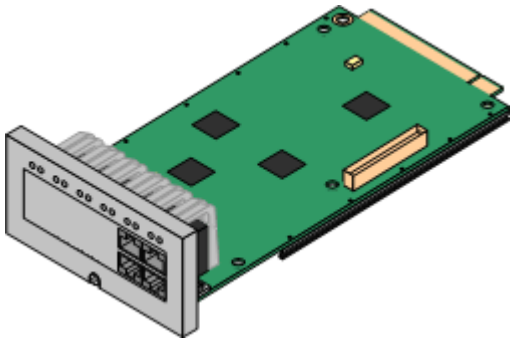
- 2 BRI trunk ports (4 channels).
- 2 Analog extension ports for the connection of analog telephones.
- 6 Digital Station (DS) ports for the connection of Avaya digital telephones
- 10 VCM channels to support IP telephones or voice networking



This card already includes an IP500 trunk card, therefore no additional trunk card is possible. it provides a single power-fail port (either port 7 or port 8). A maximum of two of these cards is allowed in the chassis.

IP500 VCM Card

This card provides voice compression channels for use with VoIP calls, SIP trunks and IP-based voice networking. The module is available in variants supporting 32 or 64 channels.



The maximum number of voice compression channels supported, using IP500 VCM base cards, Combination Cards and / or IP400 VCM cards on IP500 Legacy Card Carriers, is 148.

Please note, the maximum number of channels which may be used on each IP500 VCM card varies according to the codec used, as shown in the following table.

Maximum VCM channels available, by codec type:

	VCM 32 Card	VCM 64 Card	Combination Card
G.711	32	64	10
G.729	30	60	10
G.723	22	44	10

- This card accepts one IP500 trunk card of any type.

For more details about VCM channel requirements please refer to IP Telephony 232.

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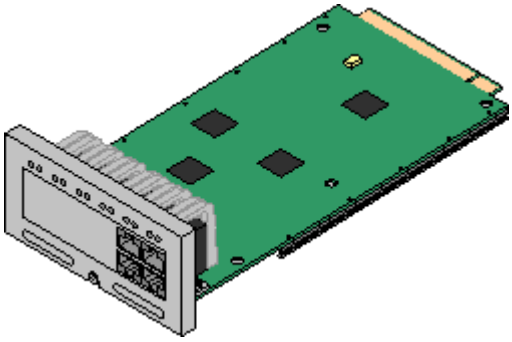
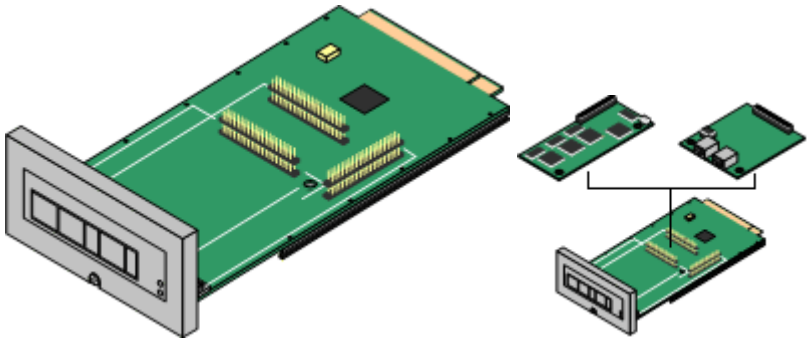
IP500 4-Port Expansion Card

- This card provides connectors for 4 additional expansion modules.
- The 4-port expansion card must be inserted in slot 4 of the control unit.
- It is not possible to connect a trunk card to the 4-port expansion card.



IP500 Legacy Card Carrier

This card allows a variety of IP400 trunk or VCM cards to be used with the IP500 control unit. The front of the card includes a number of panels that can be snapped off to match the ports available when an IP400 trunk card is fitted.



- This card does not accept any IP500 trunk cards.
- The IP500 control unit can accept up to 2 IP400 trunk or VCM cards by mounting each card on an IP500 Legacy Card Carrier
- This card supports the following IP400 cards:
  - ✓ PRI T1
  - ✓ Dual PRI T1
  - ✓ PRI 30 E1 (1.4)
  - ✓ Dual PRI E1
  - ✓ PRI 30 E1R2 RJ45
  - ✓ Dual PRI E1R2 RJ45
  - ✓ ANLG 4 Uni
  - ✓ BRI-8 (UNI)
  - ✓ VCM 4
  - ✓ VCM 8
  - ✓ VCM 16
  - ✓ VCM 24
  - ✓ VCM 30

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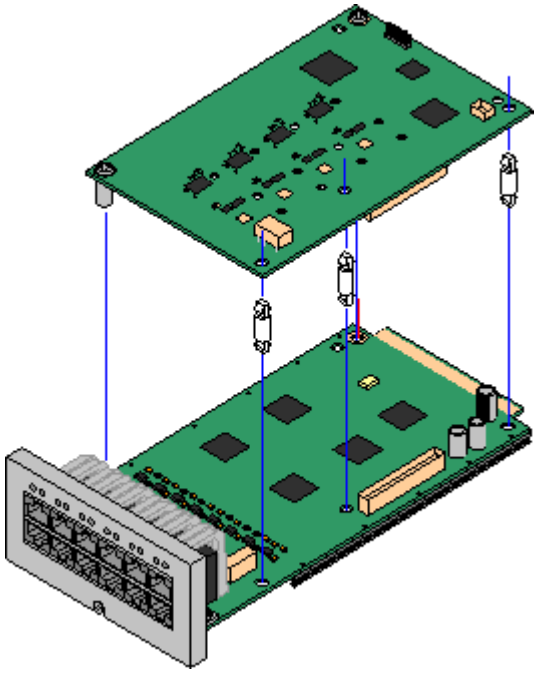
IP500 Trunk Cards

IP500 trunk cards can be fitted to existing IP500 base cards to provide support for trunk ports. The trunk card uses the ports provided on the base card for cable connection. The addition of an IP500 trunk card is supported on IP500 Digital Station, IP500 Analog Phone and IP500 VCM base cards. They are not supported on the IP500 Legacy Card Carrier base card.



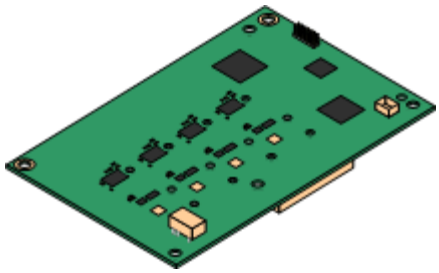
For those base cards that support trunk cards, there are no restrictions on the combination of card types. However in systems with both Analog Phone 8 base cards and analog trunk cards, combining the two types are recommended as it then provides analog power failure support for one trunk/extension (Not applicable to the Analog Phone 2 base card).

Each trunk card is supplied with the spacer pegs required for installation and a label to identify the card's presence on the physical unit once installed.

- IP500 Analog Trunk Card *(Maximum 4).*
- IP500 BRI Trunk Card *(Maximum 4).*
- IP500 Universal PRI Trunk Card *(Maximum 4).*

IP500 Analog Trunk Card

This card can be added to an IP500 Digital Station card, IP500 Analog Phone base card, or IP500 VCM card. It allows that card to then also support 4 analog loop-start trunks. Only one analog V.32 modem is supported on a system.



- When fitted to an IP500 Analog Phone 8 base card (not Phone 2), the combination supports 1 power failure extension to trunk connection.

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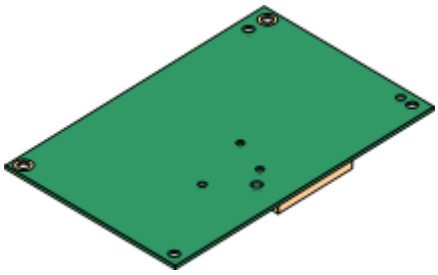
IP500 BRI Trunk Card (Euro ISDN)

This type of card can be added to an IP500 Digital Station card, IP500 Analog Phone card, TMC8 card or IP500 VCM card. It allows that card to then also support up to 4 BRI trunk connections, each trunk providing 2B+D digital channels. The card is available in 2 port (4 channels) and 4 port (8 channels) variants.

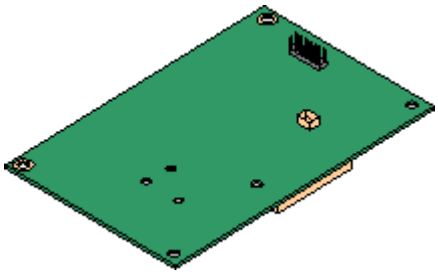


The ports on IP500 BRI cards can be individually configured to operate as an S-interface (S0 mode). When set as an S interface the settings for the Line will be exactly the same as for a circuit on an So8 Expansion Module.

- These trunks support the mobility features of Mobile Call Control and one-X Mobile client 210.

IP500 Universal PRI Trunk Card

This type of card can be added to an IP500 Digital Station card, IP500 Analog Phone card, TMC8 card or IP500 VCM card. It allows that card to then also support primary rate digital trunk connections. Available in single and dual versions the IP500 PRI card provides single and dual primary rate trunk interfaces respectively. The PRI is configurable for T1, E1 or E1R2 MFC use depending on the territory.



Details of the supported ISDN supplementary services and protocols for each PRI are given in the 'Public and Private Voice Networks' section.

The IP500 Universal PRI trunk cards incorporate an integrated CSU/DSU. The CSU function allows the trunk to be put in loop-back mode for testing purposes. This can be set manually, using the monitor application, or automatically from a Central Office sending a Line Loop Back (LLB) pattern. The DSU function allows the T1 trunk to be shared between data and voice services.

Here is a summary of the capabilities of the card:

- Each card is configurable to connect to T1, E1 or E1R2 lines.
- The card is available in either a single or dual PRI variant. The single variant can support up to 24 T1 channels or up to 30 E1 channels. The dual variant can support up to 48 T1 channels or 60 E1 channels.
- On each card, 8 channels per interface are enabled by default. This means that the single PRI has 8 channels enabled, while the dual PRI has 16 channels enabled. Further channels may be enabled by the purchase of additional licenses in 2-channel or 8-channel increments.
- The IP500 PRI trunk card works on any IP500 VCM or extension base card (not the Legacy Card Carrier).
- Up to four Universal PRI cards can be installed in any combination in the IP500 or IP500 V2 chassis.
- Diagnostics capabilities:
  - Visual indicators to show service state
  - Physical test points to monitor traffic.
- These trunks support the mobility features of Mobile Call Control and one-X Mobile client 210.

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External Expansion Modules

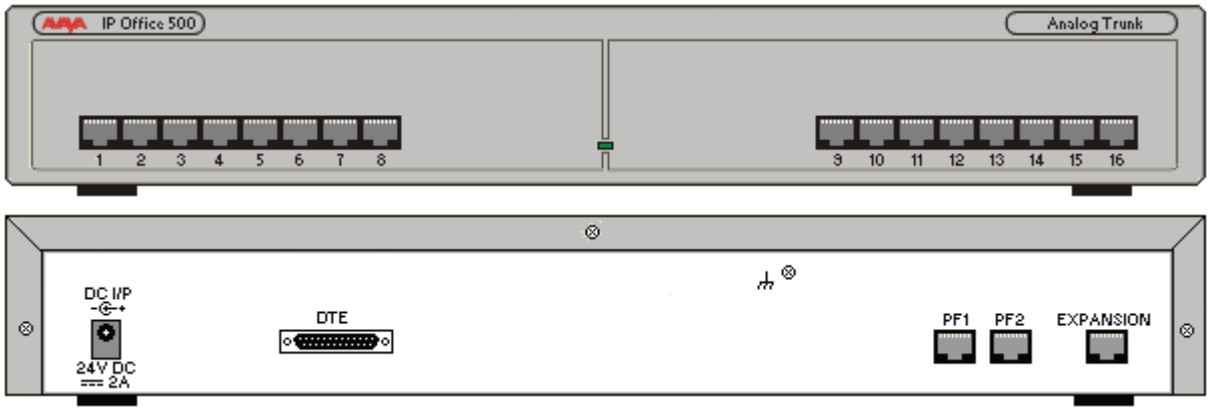
Each of these modules may be used with the IP500

- IP500 Phone Expansion Module 95
  Available in two variants for 16 or 30 analog extensions with calling line presentation.
- IP500 Digital Station Expansion Module 94
  Available in two variants for 16 or 30 digital extensions for Avaya series digital telephones or select Nortel digital telephones (Digital Station 16A and Digital Station 30A variants used with Nortel telephones with IP Office Release 7.0)
- IP500 Analog Trunk 16 Expansion Module 93 *(US version only)*
  Provides 16 analog loop start or ground start trunks, with power failover of two trunks.

IP500 Analog Trunk 16 Module

This expansion module provides an additional sixteen Loop Start or Ground Start two-wire analog trunks. (Ground start trunks are not available in all territories) The first two trunks on the module which are automatically switched to power fail sockets on the rear of the unit in the event of power being interrupted must be loop start for correct power fail operation.

For installations in a rack, this module requires the IP500 Rack Mounting Kit. The IP500 Analog Trunk 16 Module is functionally identical to the IP400 Analog Trunk 16 Module.



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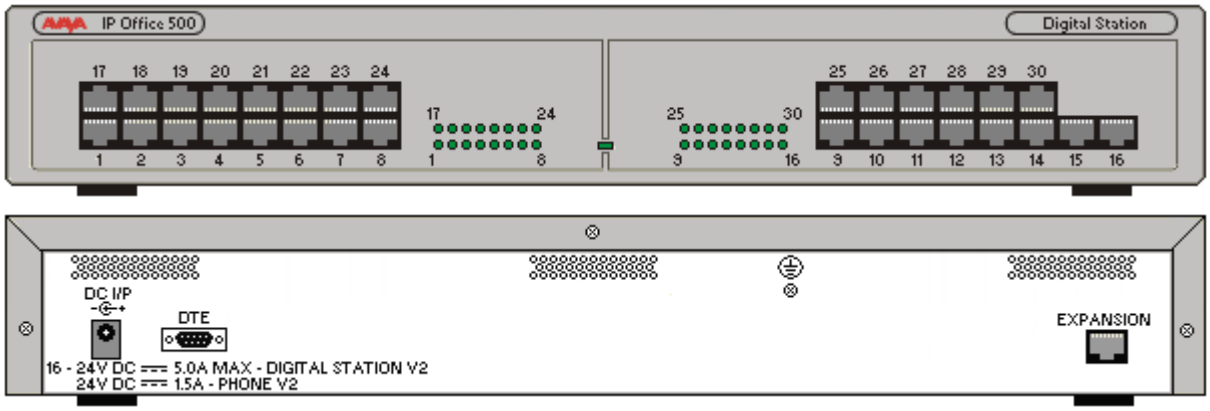
IP500 Digital Station Module

This expansion module provides additional Digital Station (DS) ports for selected Avaya 1400, 2400, 4400, 5400, 9400, 9500, 6400, T3 (EMEA only) Series telephones and 3810 wireless telephones (North America only). The IP500 Digital Station module is available in several variants; 16 or 30 extensions, supporting IP Office telephones with RJ45 connectors and supporting Nortel telephones with RJ21 connectors. Refer to the Telephone section for Nortel phone support.

For installations in a rack, this module requires the IP500 Rack Mounting Kit. The IP500 Digital Station Module is functionally identical to the IP400 Digital Station V2 Module.



- Telephones can be located up to 3280 feet (1km) from the control unit. For extensions located "out-of-the-building" additional line protection will be needed. For more information on cabling and out of building guidelines, see the IP Office Installation Manual.
- For systems where Direct Station Select (DSS) Units are being used, IP Office supports a maximum of:
  - BM32: *(1616)*
    Up to 32 DBM32s supported (32 buttons each, up to 1024 buttons).
  - DSS4450: *(4412D+, 4424D+)*
    Up to 8 DSS4450 modules per system (maximum of 2 per DS expansion module) (50 buttons each, max 400 buttons).
  - DBM32: *(1416)*
    Up to 32 DBM32s supported (32 buttons each, up to 1024 buttons).
  - BM12: (9508 digital; 9608 and 9641 IP telephones)

    Up to 36 buttons supported per telephone (3xBM12), up to 1024 buttons per system) system
  - EU24/EU24BL: *(4620, 4621, 5420, 5620, 5621/4620, 4621, 5620, 5621)*
    Up to 8 EU24s per system (24 buttons each, max 196 buttons).
  - SBM24: *(9630G, 9640, 9640G, 9650, 9650C)*
    Up to 42 SBM24 per system (24 buttons each, up to 1008 buttons).
  - T3 Series: *(All T3 telephones)*
    Up to 30 T3 DSS modules supported. (36 buttons each, up to 1080 buttons). T3 DSS buttons are not included in the overall system limit for additional button module buttons.

See the Telephones Section 98 for specific limits on the number of each type of telephone supported on DS modules.

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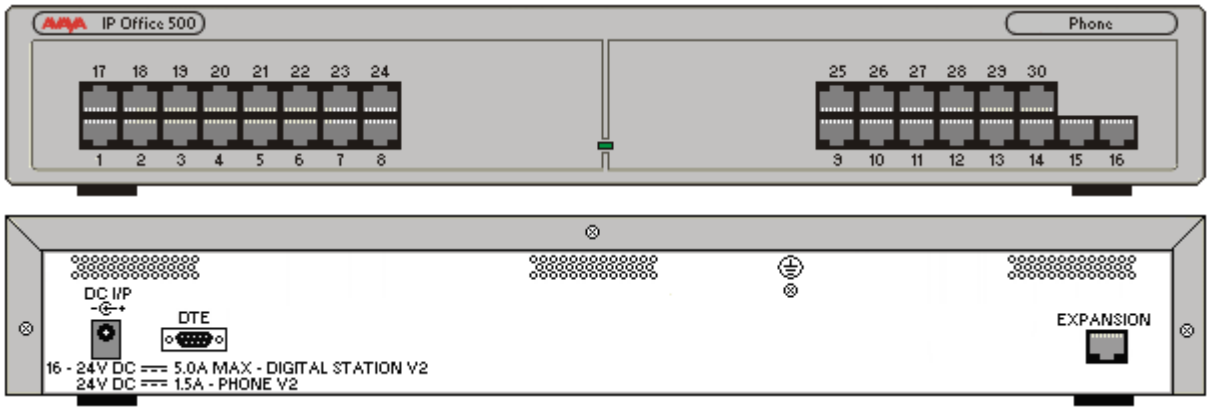
IP500 Phone Expansion Module

This module provides additional analog telephone interfaces:

- Two Wire.
- DTMF signaling (No rotary or Loop Disconnect).
- Time Break Recall (No Earth Recall).
- Caller ID capable.
- Message Waiting Indication (MWI) capable - High Voltage, Pulsed High Voltage, Line Reversal.

The IP500 Phone module is available in 2 versions, giving 16 or 30 extensions. Telephones can be located up to 3280 feet (1km) from the control unit. For extensions located "out-of-the-building" additional line protection will be needed. For more information on cabling and out of building guidelines, see the IP Office Installation Manual.



- IP Office Phone Modules provide support for a variety of analog MWI methods. These methods are 51V Stepped, 81V, 101V and Line Reversal.
- Each analog port can support a device of maximum 1 REN.
- On analog ports, call information is sent while the phone is ringing, and cannot be updated during a call or set on an outbound call (the phone may do a local match but this is not controlled by the IP Office). The primary purpose of displays is to give information about incoming calls. Where the Caller Display standard chosen supports the delivery of text (extension name) as well as the number, both are delivered.
- An analog extension port can be set for external Paging operation. It does not operate like a normal extension and is connected to external equipment through an isolation device. The Port will always be busy so it cannot be called directly and can only be accessed by using a paging feature. When not receiving a Page, the port will remain silent. When being paged, the page tone is sent before the speech path is opened.

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Chapter 5. Telephones

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5. Telephones

The right phone for the User:

With a wide choice of desktop telephones, supporting both digital and IP interfaces, Avaya IP Office has always the right phone for a user. The portfolio is further enhanced with solutions for in-building wireless mobility, both using DECT and WIFI technology as well as The IP Office Video Softphone.

There is an easy way to select the right phone for your customer:

1) Decide on phone line: Cost efficient value line, key centric with paper labels? Or easy to maintain self labeling telephones, for customers where low TCO is important and installation and upgrading cost are a concern

2) Decide on Interface: Forward looking IP telephones for a Voice ready IP network? Or more traditional digital telephones, e.g. twisted pair cabling? Avaya gives the choice, while maintaining a consistent user interface and consistent functionality across both lines.

3) Decide in User profile: What role does the user have? Is he an everyday user, needing a standard phone workhorse? Or does he – e.g. as a manager – have to handle many calls? The User profiles give an easy way to get the right phone in each category for each person.

Key-driven telephones for the cost-sensitive customer

The 1600 IP phone and 1400 digital telephones are classical, cost effective Telephones designed especially around the needs of the Small and Mid-Size Business customer. Keys with clear red and green LEDs make access to lines and feature a breeze. These telephones are especially well suited for cost sensitive customers as well in scenarios where “key-system-functionality” is needed. They provide an easy migration path for users that have been accustomed to e.g. the 4400 line or the PARTNER line of digital telephones.

Paperless telephones for fast installation and cost efficient changes:

The new 9600 IP telephones, as well as the 9500 Digital telephones are especially welcomed where a low cost of ownership is important. All telephones are equipped with paperless keys, making it easy to change features and button assignments from remote. Shift/page button allows access to a large number of lines or features that are shown dynamically in the display, making access to IP Office functionality easy while providing a clean, simple to use user-interface.

Speakerphone options

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Full-duplex speakerphone options are available for people that need a high quality personal speakerphone. All speakerphones are for personal use for one or two people sitting in front of the phone, not for a group-conferencing with people surrounding the phone.

Access to telephones everywhere in and IP Office Network (SCN):

With support of hotdesking across a large line of telephones, IP Office telephones are perfect in an environment where people change their workplace regularly. With a simple login, users can make any telephone, digital or IP, their personal extension. This works not only in a single IP Office but also across a Multi-Site Network (Small Community Network), making traveling from one company site to another a breeze.

Even better, on 9600, 1600 and 1400 telephones, this functionality has been further enhanced by providing a centralized call log, directory and personal settings. When hotdesking to a different phone, call-logs and directory are instantly updated and synchronized over the network, eliminating e.g. lost calls as the call log was not available at a remote site.

The right phone for every user:

Avaya IP Office Desktop telephones are designed to meet typical user needs in the small and mid business market:

- Walk Up
  Telephones in common areas like a meetings room, lobby or store. The telephones are normally used by casual users that walk-up to a phone where they make a quick phone call. Normally, no sophisticated features are needed.
  - Example user: Customer, Visitor, Meeting room phone.
- Everyday
  Users that need a phone on their desk as part of their everyday job. While having a phone is important, the phone is just another tool on the desk for users using several alternative communication technologies.
  - Example user: Engineer, Accountant.
- Receptionist/Assistant
  A typical user manages a lot of calls, often as a significant part of their job. In many cases they also manage calls for others users and need to be efficient in connecting callers to the right people.
  - Example user: Executive Assistant, Receptionist.
- Executive
  For these users, handling a lot of calls is essential and they are constantly on the phone. Quite often, they handle multiple calls; they must always be connected, even when mobile. They require a set of rich, yet easy to use communication capabilities.
  - Example user: Sales Executive, Banker, Lawyer, Project leader.

telephones for the different Usage categories:

- Walk Up: 1603, *Digital:*1403
- Everyday: *IP* 1603SW (Basic Everyday), 1608, 9608, *Digital:*1408, 9504, 9508
- Receptionist/Assistant: : *IP* 1616 optional with BM32, 9608 optional with BM12; 9621, 9641 with optional BM12, *Digital:* 1416 with optional DBM32 9508 with optional BM12,
- Executive: *IP* 9621, 9641, *Digital:* 9508 with optional BM12

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9600 Series IP Telephones

Avaya one-X Deskphone is a family of next generation IP telephones that delivers a new and unique communications experience to drive increased productivity.

The Avaya one-X Deskphone family features an intuitive user interface which helps to make users proficient and confident in performing common telephone tasks such as setting up a conference call or completing a transfer. With high quality audio it is much easier to hear and understand other people, which speeds business while reducing fatigue and stress. The Avaya 9600 Series Deskphones are built with future growth and enhancement in mind, with many modular add-ons that can be added as they are needed – protecting investments and leading to improved total cost of ownership. The new telephones also feature a very stylish and professional design including support for customized display screen saver images and custom faceplates with company logos.

Common to all phones of the 9600 line on IP Office are two access modes to important IP Office features: Features can be assigned to the line/feature key of the telephone for easy and quick one-touch selection of the functionality giving users of other Avaya telephones a consistent user-interface.

In addition the features can be accessed through a menu, allowing quick access to seldom used features or even functionality that is not administered to a key. Similar to menus found on mobile and smartphones, this intuitive, “self-service” access frees the user from limits imposed by the number of keys on a telephone and allows access to the vast functionality of IP Office.

The status indication presents a clear view of important call settings like call forwarding numbers.

Visual Voice gives menu access to the voicemail messages received by a user for quick retrieval of the most critical information.

Personal and centralized directory access makes calling a breeze. A centralized call log allows one-touch call-back of missed calls, and is fully synchronized with other Avaya IP telephones or the one-X Portal application.

Full user mobility between different 9600 telephones, 1600 telephones and 1400 digital telephones support hotdesking and user mobility between different telephones with access to all the features found on the ‘home’ phone.

Customers are becoming more aware of the need to conserve energy due to costs and environmental considerations. The 9600 series telephones have best in class power performance with the 9620L meeting PoE Class 1, and all other models being Class 2, including the Gigabit versions. Over the life of a product the reduced power demand of the 9600 series results in significant cost savings for customers when compared to other competitive offerings that require more power, such as PoE Class 3 devices.

Please note that the 9600 telephones as well as the 9500 telephones are supported on IP Office IP500 and IP500 V2 hardware only!

9608 and 9611G

Note: The 9611 telephones will not be available at launch of IP Office Release 7.0 but will be made available at a later time. Please check availability before offering to avoid misunderstandings.

The two button-orientated models, the 9608 and 9611G, provide a more traditional user experience with dual color buttons on either side of the graphical displays. These are ideal for customers who are migrating from a traditional digital environment and desire to move to IP with minimal change. As well as targeting more traditional users, these models are suited for Everyday users who use voice as an element within their business communications but are not considered power users.
The 9608 and 9611G models have 8 buttons around the display with 4 on the left and 4 on the right. Users can select to display information in a single column with four rows or in two columns with four rows each, depending on their needs. In two-column mode, approximately thirteen characters can be displayed on a line in each column. The exact number depends on the language and the characters because the font includes variable width characters.
Avaya continues to focus on low power and these two models are both Power over Ethernet Class 1 (802.3af). These telephones retain the industrial design of the other 9600 Series models. Please note that the power requirement and PoE class of the sets varies based on the configuration of connected interfaces such as button modules.
Both telephones are ready for future support of Wideband Audio on IP Office, giving a future proof investment path. Both models include full duplex hands-free support with a speaker and microphone.
The 9608 and 9611G include support for a new 2 x 12 button module on which features, speed dials, bridged or individual call appearances can be programmed. The current 24 button module is also supported, but the 12 and 24 button modules cannot be intermixed when connected to a single phone. A maximum of 3 button modules of either type can be connected to these Deskphones.
Each model also includes one touch access to key applications including Contacts, History and Home. One touch access to these core applications simplifies the user experience and eliminates the need to navigate through menus to get to the most commonly used features.
On top of that, when connected to IP Office these telephones will offer access to advanced IP Office features like Visual Voice Mail, IP Office Feature menu or centralized storage for call log, system directory and personal directory. With that they offer full support of the hot-desking capabilities of IP Office.
The 9611G includes a color screen. The 9608 has a monochrome display.

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These models support a dual position stand at approximately 40□ or 60□. The display is fixed and is not adjustable like other 9600 Series IP Deskphones. The 9608 and 9611G do not support removable faceplates.

Product Overview:

	9608	9611G
Buttons	8	8
Administrable buttons (3 pages)	24	24
Color	No	Yes
Display (fixed)	3.8 inch / 9.7 cm	3.5 inch / 8.9 cm
Softkeys	4	4
Dual Position Stand	Yes	Yes
Integrated Switch	10/100	10/100/1000
USB Interface	No	Yes
Wideband Capability	Handset, headset	Handset, headset
Speaker	Yes	Yes
PoE Class (base level)	1	1
Applications Buttons	Contacts, Home, History, Message, Phone	Contacts, Home, History, Message, Phone
Other Keys	Volume, Mute, Headset, Speaker	Volume, Mute, Headset, Speaker
5 Way Navigation Cluster	Yes	Yes
Customizable Faceplate	No	No
Maximum Button Modules (12 and 24 buttons)	3	3

Note: Wideband currently not supported when telephones are used with IP Office:

	IP Office (IP500/IP500 V2)	Communication Manager
9608	✓	✓
9611	✗(in prepar.)	✓

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Common Features:

- Display: 9608: Monochrome display – 3.2 inches x 2.2 inches (8.2 cm x 5.5 cm); 9611: Color display 2.8 inches x 2.1 inches (7.0 cm x 5.3 cm)

  Fixed Feature Buttons: 10 plus Four-way navigation cluster button.

  - Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED), Speaker button(LED), Headset button(LED).
  - Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen.
  - Contacts button, history (Call log) button (LED), quick-access voicemail Message button with LED and corner message indicator.
- Programmable /Contextual buttons:
  - 24 - in 3 switchable display pages of 8 matching the 8 physical display buttons. All 8 buttons with Dual LED indicator
  - 4 contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
- Key Labels: Icons and text used on fixed feature keys.
- Speakerphone: Full duplex speakerphone.
- Hearing Aid Compatible: Yes.
- Message Waiting Indicator: Yes - can also be used as a ringing call alert indicator.
- Personalized Ring Patterns: Three distinct ringtone styles. These include Classic, Alternate and Rich. The Classic and Alternate versions provide more legacy sounding ring tones. The Rich ringtones are more elaborate and nontraditional..
- Headset Socket: Yes. Bluetooth and DECT headset support (with adapter)
- Embedded Applications:
  - Centralized call log and contact application, fully transparent between 9600 telephones, 1408/1416/1608/1616 telephones and one-X Portal for IP Office.
  - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls). Please note that the Voice-dial feature of the 9600 telephones on Communication is not available on IP Office.
  - Access to company directory and Visual Voice applications on IP Office.
- Upgradeable Firmware: Yes.
- Expansion:. yes. Support of up to 3*BM12 button (2*12 buttons per module, total of max. 64 available buttons)
- Color: gray.
- Mounting: Desk or wall mountable.
- Adjustable Desk Stand: Yes two-position stand, approx 40° and 60°angle
- IP-Details:
- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit
- Power Class: PoE Class (IEEE 802.3af) registers as class 1 device
- Codecs: G.711, G.729a/b. ready for G.
- Supported Standards including Quality of Service:
  - TCP/UDP/RTP/RTCP/SRTP
  - RSVP
  - LLDP/LLDP-MED (inc VLAN assignment)
  - ARP/DNS
  - 802.1Q (Layer 2 QoS)
  - VLAN Separation
  - DiffServ (Layer 3 QoS)
  - Port Range (Layer 4 QoS)

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- 802.1X (MD-5)

- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports:

  9608: Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.

  9611: Two port full-duplex 10/100/1000 BaseT Ethernet switch for PC pass-through connection.

9621G and 9641G

The color touch-screen models, the 9621G and 9641G, evolve the user experience of the 9600 Series with a touch interface. Having color touch-screens on more models, with flexible software that can be enhanced in the future, these Deskphones provide a platform for near and long term enhancements that leverage the touch-screen capabilities. The 9621G and 9641G are targeted at Essential users who rely on the phone for business communications or those users that can benefit from tighter and easier access to a wider range of Avaya applications.
Both telephones are ready for future support of Wideband Audio on IP Office, giving a future proof investment path. The full duplex hands free audio performance is truly differentiating for heavy speaker phone users.
All the new 9600 Series IP Deskphones are low power including the 9621G and 9641G which are both PoE Class 2. The 9641G is the first IP Deskphone that can provide Gigabit Ethernet, a color screen AND a button module while consuming only PoE Class two power.
The 9621G and 9641G have a similar look and feel to the current 9600 series models and include a dual position flip stand that positions the phone at approximately 20□ in the low position or 60□ in the upright position. The high resolution touch-screen displays are adjustable to 4 independent positions and both models support removable faceplates and have separate part numbers for versions without a faceplate.
The 9641G can be used with the new 2 x 12 button module and the original 24 button version. A maximum of three button modules can be connected at one time. Please note that the power class will be increased with the addition of button modules. The 12 button and 24 button modules cannot be intermixed on a single phone.
The 9621G does not support a button module.
The 9641G also includes a USB interface (v2.0).

Product Overview:

	9621G	9641G
Touch Screen	Yes	Yes
Administrable buttons (3 pages)	24	24
Color	Yes	Yes
Display (fixed)	4.3 inch /10.9 cm	4.7 inch / 11.9 cm
Dual Position Stand	Yes	Yes
Integrated Switch	10/100/1000	10/100/1000
USB Interface	No	Yes
Wideband Capability	Handset, headset, speaker	Handset, headset, speaker
PoE Class	2	2

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Applications Buttons	Contacts, Home, History, Message, Phone, Forward	Contacts, Home, History, Message, Phone, Forward
Other Keys	Volume, Mute, Headset, Speaker	Volume, Mute, Headset, Speaker
Customizable Faceplate	Yes	Yes
Bluetooth Interface	No	Yes
Maximum Button Modules (12 and 24 buttons)	0	3

Note: Wideband currently not supported when telephones are used with IP Office:

	IP Office (IP500/IP500 V2)	Communication Manager
9621	✓	✓
9641	✓	✓

Common Features:

- Display:
  - 9621: Color touch-screen display – 3.7 inches x 2.1 inches (9.5 cm x 5.4 cm);
  - 9641: Color touch screen display 4.1 inches x 2.3 inches (10.4cm x 5.9cm)
  - 4-position adjustable tilt display

  Fixed Feature Buttons: 11
  - Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED), Speaker button(LED), Headset button(LED).
  - Avaya Menu button (browser, options and settings access), Home/Telephony application button – to return to main telephone screen.
  - Contacts button, history (Call log) button (LED), quick-access voicemail Message button with LED and corner message indicator. Call Forward button
- Programmable /Contextual buttons::
  - 24 administrable keys/functions accessible through touch screen (different layout options)
  - 0–5 contextual softkey buttons (touch screen). Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
- Key Labels: Icons and text used on fixed feature keys.
- Speakerphone: Full duplex speakerphone.
- Hearing Aid Compatible: Yes.
- Message Waiting Indicator: Yes - can also be used as a ringing call alert indicator.
- Personalized Ring Patterns: Three distinct ringtone styles. These include Classic, Alternate and Rich. The Classic and Alternate versions provide more legacy sounding ring tones. The Rich ringtones are more elaborate and nontraditional..
- Headset Socket: Yes. Bluetooth and DECT headset support (with adapter); 9641: built in Blueooth adapter.
- Embedded Applications:
  - Centralized call log and contact application, fully transparent between 9600 telephones, 9500 telephones, 1408/1416/1608/1616 telephones and one-X Portal for IP Office.
  - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls). • Access to company directory and Visual Voice applications on IP Office.
- Upgradeable Firmware: Yes.
- Expansion: yes. Support of up to 3*BM12 button module (9641 only).(2*12 buttons per module)
- Color: gray.

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- Mounting: Desk or wall mountable.
- Adjustable Desk Stand: Yes Reversible wedge stand for desktop or wall-mount use and dual-position flip stand

IP-Details:

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit
- Power Class: PoE Class (IEEE 802.3af) registers as class 2 device
- Codecs: G.711, G.729a/b. ready for G.
- Supported Standards including Quality of Service:
    - TCP/UDP/RTP/RTCP/SRTP
    - RSVP
    - LLDP/LLDP-MED (inc VLAN assignment)
    - ARP/DNS
    - 802.1Q (Layer 2 QoS)
    - VLAN Separation
    - DiffServ (Layer 3 QoS)
    - Port Range (Layer 4 QoS)
    - 802.1X (MD-5)
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports:

  Two port full-duplex 10/100/1000 BaseT Ethernet switch for PC pass-through connection.

9504 and 9508 digital phones for IP Office

Please note that the 9600 telephones as well as the 9500 telephones are supported on IP Office IP500 and IP500 V2 hardware only

Avaya is continuing driving design innovation by offering two new paperless digital sets in the same design line as the 9600 telephones:
These telephones are delivering all the core values of the 5400 digital sets while innovating in areas like backlight displays for easy readability and full duplex handsfree support also on digital sets (9508 only)
The two button-orientated models, the 9504 and 9508, provide a traditional user experience with dual color buttons on either side of the graphical displays. These are ideal for customers who are migrating from a traditional digital environment and desire to move to IP with minimal change. As well as targeting more traditional users, these models are suited for Everyday users who use voice as an element within their business communications but are not considered power users.
The 9504 and 9508 models have 4/8 buttons around the display with 2/4 on the left and 2/4 on the right. Users can select to display information in a single column with two/four rows or in two columns with two/four rows each, depending on their needs. In two-column mode, approximately thirteen characters can be displayed on a line in each column. The exact number depends on the language and the characters because the font includes variable width characters.
As digital telephones, the new sets offer superior power efficiency. They are full powered from IP Office when
The 9508 includes support for a new 2 x 12 button module on which features, speed dials, bridged or individual call appearances can be programmed. A maximum of 3 button modules of either type can be connected to these Deskphones. Please note that a separate power supply is needed when a BM12 is connected.
Each model also includes one touch access to key applications including Contacts, History (previously Call Log) and Home (previously menu). One touch access to these core applications simplifies the user experience and eliminated the new to navigate through menus to get to the most commonly used features.
On top of that, when connected to IP Office these telephones will offer access to advance IP Office features like Visual Voice Mail, IP Office Feature menu or centralized storage for call log, system directory and personnel directory. With that they offer full support of the hot-desking capabilities of IP Office.
The 9504 has a four-line monochrome display with backlight, while the 9508 has the same backlight 8-line monochrome display as the 9608 IP version.
These models support a dual position stand at approximately 40□ or 60□. The display is fixed and is not adjustable like other 9600 Series IP Deskphones. The 9504 and 9508 do not support removable faceplates.

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Product Overview:

	9504	9508
Physical Buttons	4	8
Administrable buttons (3 pages)	12	24
Number of display Lines	4	8
Display (fixed)	4-line .	3.8 inch / 9.7 cm
Softkeys	4	4
Dual Position Stand	Yes	Yes
Speaker/handsfree	Yes	Yes. DSP based full duplex handsfree
Applications Buttons	Contacts, Home, History, Message, Phone	Contacts, Home, History, Message, Phone
Other Keys	Volume, Mute, Headset, Speaker	Volume, Mute, Headset, Speaker
5 Way Navigation Cluster	Yes	Yes
Customizable Faceplate	No	No
Maximum Button Modules (2*12)	none	3

	IP Office (IP500/IP500 V2)	Communication Manager
9504	✓	✗
9508	✓	✗

Note: a second version of these digital telephones is planned for launch with Communication manager at a later time.

Common Features:

- Display: 9508: Monochrome Backlit display – 3.2 inches x 2.2 inches (8.2 cm x 5.5 cm); 9504: Monochrome Backlit display – 4 rows by 24 characters

Fixed Feature Buttons: 10 plus Four-way navigation cluster button.

- Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED), Speaker button(LED), Headset button(LED).
- Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen.

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- Contacts button, history (Call log) button (LED), quick-access voicemail Message button with LED and corner message indicator.

- Programmable /Contextual buttons:
  - 9508: 24 - in 3 switchable display pages of 8 matching the 8 physical display buttons. All 8 buttons with Dual LED indicator
  - 9504: 12 - in 3 switchable display pages of 4 matching the 4 physical display buttons. All 4 buttons with Dual LED indicator
  - 4 contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
- Key Labels: Icons and text used on fixed feature keys.
- Speakerphone: Full duplex speakerphone (9508), Two-way speakerphone (9504)
- Hearing Aid Compatible: Yes.
- Message Waiting Indicator: Yes - can also be used as a ringing call alert indicator.
- Personalized Ring Patterns: Yes – 8 (Traditional Avaya)
- Headset Socket: Yes. Bluetooth and DECT headset support (with adapter)
- Embedded Applications:
  - Centralized call log and contact application, fully transparent between 9600 telephones, 1408/1416/1608/1616 telephones and one-X Portal for IP Office.
  - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls). Please note that the Voice-dial feature of the 9600 telephones on Communication is not available on IP Office.
  - Access to company directory and Visual Voice applications on IP Office.
- Upgradeable Firmware: Yes.
- Expansion:. yes. Support of up to 3*BM12 button (2*12 buttons per module, total of max. 64 available buttons) (9508 only). A separate power supply is needed when connection the BM12 to the phone as the digital line can not power phone plus button module
- Color: gray.
- Mounting: Desk or wall mountable.
- Adjustable Desk Stand: Yes two-position stand, approx 40° and 60°angle

Button Module 12 for 9508, 9608, 9641

A new button module is being added to the portfolio. The 2 x 12 Button Module provides dual color LEDs/buttons aligned with 12 rows on the graphical display. It includes two pages with 12 buttons each and a left or right arrow to indicate which page is currently not visible. When both pages are programmed the user has a total of 24 buttons available.
The 12 button module also includes a new quick-labelling feature button that is located under the display. Three 12 button module can be simultaneously connected to any one of the new 9600 Series IP Deskphones except the 9621G as well as with the 9508 digital phone. This button module ships with the same dual position stand as the 9508/9608

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	Up to three modules to be connected
9608	✓
9611	✓
9621	✗
9641	✓
9504	✗
9508	✓

- Programmable /Contextual buttons:
  - 24 administrable keys/functions accessible
  - 12 physical keys with LCD labels and dual LED
  - Quick-labeling key plus left/right page-key
- Color: gray.

  Note: when 9508 is used, a separate power supply is needed to power the module.

1600 Series IP Telephones 1400 Series Digital Telephones

Avaya one-X is a portfolio of communications solutions that deliver a powerful and consistent communications experience for the end user – across a variety of devices and interfaces. Avaya one-X solutions provide Intelligent Access to Intelligent Communications driving enhanced productivity and competitive advantage.

Avaya one-X Deskphone Value Edition, also called the 1600 Series of IP telephones, is a family of cost effective IP Telephones designed especially around the needs of the Small and Mid-Size Business customer.

For users of a digital solution, the 1400 Series of Digital telephones offer the same functionality and consistent communication experience on a line of digital telephones, giving customers the choice to select the right solution regardless of the method of connection.

With IP Office Release 6, the line of 1600 telephones has been further enhanced with the 1600-I series, providing an easier-to-read graphical display with white backlight, consistent with the displays of the 1400 family.

Designed with the reliability you expect from Avaya, one-X Deskphone Value Edition telephones provide critical features and capabilities not often found in competitively priced models.

The family of 1400/1600 telephones consist of three models, designed around different user needs:

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- The 1403/1603 phone is designed for the Walk Up user and basic Everyday user. For a Walk Up scenario, the 1403/1603 is used common areas in offices, stockrooms, lobbies, or drop-in desks. Users can be building visitors, employees, or even customers who need a phone with a simple, familiar user interface. The 1603 telephones comes in two versions: The 1603, a basic phone without Ethernet switch for locations where no PC is connected to the phone, typically for the walkup user, as well as the 1603 SW, with a basic Ethernet switch to connect additional components to the phone, well suited for the basic Everyday user.
- The Avaya 1408/1608 IP Telephone is designed for the Everyday user. Everyday users typically rely on several forms of communication including voice and email – and while they require a quality telephone, they rarely handle large amount of calls at the same time. Cubicle workers and sales staff are examples of Everyday users for whom the 1408/1608 delivers a productivity-enhancing telephone.
- The Avaya 1416/1616 IP Telephone is designed for the Receptionist/Assistant user. Receptionists, secretaries, and managers are examples of this type of user – people who answer incoming calls, transfer customers to different departments or extensions, and monitor several line appearances throughout a typical day. For the Receptionist/Assistant user, the 1416/1616 provides the most one-touch line/feature/speed-dial buttons without the need to scroll through on-screen lists.

All 1400/1600 telephones share a consistent design line and feature a button-centric layout. With fast access to all needed features like hold, transfer and conference, easily readable back-lit displays and a stylish design, they are an important addition to the Avaya portfolio of telephones for the Small and Mid-Size market.

All telephones share a number of feature keys with clear, paper labels. On the 1408/1608 and 1416/1616, features are also available through a feature list on screen that makes selecting features and line appearances easy even when hotdesking to another phone.

The telephones are accompanied by a complete set of accessories like cost efficient power supplies, easy to use paper labels and spare parts. For powering options for the 1600 IP telephones refer to the section Power Options for IP Telephones 240.

For easy and efficient generation of labels, free DESI printing software is available from http://support.avaya.com (on the 1600 Phone Series page). This can be used to print all labels correctly formatted on plain paper or available DESI labels. A PDF document with correct formatting can also be downloaded. Additionally, labels can be printed in the correct format directly from IP Office Manager.

For 1600 Series telephones requiring Gigabit Ethernet support, a Gigabit Ethernet adapter is available - providing 10/100/1000 Mbps support for both the 1600 Series phone and co-located laptop or PC.

Note that no Ethernet cable is included with the 1600 Series telephones. If a cable is needed, it should be ordered separately at the required length.

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1603 and 1603SW IP Telephone 1403 Digital Telephone

This telephone is best suited for the Walkup and Basic Everyday user.

Works on:	IP Office	Communication Manager
1603	✓	✓
1603 SW	✓	✓
1603-I	✓ (R5 and later)	✓
1603SW-I	✓ (R5 and later)	✓
1403	✓	✗

On the 1600 line, only the 1603-I/603SW-I versions with improved display are available.

The Avaya 1403/1603 supports 3 line appearances. Each of the buttons includes dual LEDs (red, green) providing explicit status for the user. For a familiar look and feel, the 1403/1603 includes several fixed feature keys for common telephone tasks including conference, transfer, drop, hold, and mute. In addition, the 1403/1603 includes a 2-way speakerphone. The display on the 1403/1603 measures two lines by 16 characters and is backlit for easier viewing in all lighting conditions.

The 1603 SW telephone has the same features/functionality as the 1603 phone, it adds on top of that however a secondary Ethernet port and is therefore targeted at the basic Everyday user.

The newly introduced 1603-I/1603SW-I are functionally equal to the 1603/1603SW but feature a higher quality graphical display with white backlight, identical to the display of the digital 1403 telephone.

Common Features:

- Display/Housing:
  - Backlit display – 2 rows by 16 characters. Green backlight for 1603/1603-SW, white backlight for 1603-I, 1603SW-I, 1403.
  - Reversible wedge stand for desktop and wall-mount use.
- Fixed Buttons: 10.
  - Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button, Speaker button.
  - Avaya Menu button (options and settings access).
  - Hold button, Conference button, Transfer button, Drop button, Redial button.

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- Programmable/Contextual buttons: 3.
  - 3 line appearance key buttons – with dual LEDs (red, green) and paper labels (Printing tool available).
- Key Labels: Icons and text used on fixed feature keys.
- Speakerphone: Two-way handsfree speaker and microphone.
- Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.
- Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: No.
- Upgradeable Firmware: Yes.
- Expansion: None.
- Color: Black.
- Mounting: Desk or wall mountable.

Requirements for 1403:

- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Requirements for IP interface: (1603, 1603SW, 1603-I, 1603SW-I)

- Power Supply: Optional IEEE 802.3af Power over Ethernet (PoE) adapter (class 2) or optional wall plug local power supply.
- Codecs/audio: G.711, G.729a/b, Dynamic Jitter buffer, Echo Cancellation, Comfort Noise, Automatic Gain Control.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN).
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: 1603 - Single Ethernet (10/100) line interface. Support of optional Gigabit Ethernet Adapter.
- Ethernet Ports: 1603 SW Ethernet (10/100) line interface with a secondary 10/100 port for co-located laptop or PC with VLAN separation. Support of optional Gigabit Ethernet Adapter.
  - Well suited for everyday use, where typically smaller sized data files transferred (web, email file attachments) or less demanding data-transfer application/usage (thus less sensitive to data transfer completion times).
  - For highly demanding data traffic processing with large amounts of small packets we recommend the 1616 phone for full speed data throughput.

Language Support:

- 1403/1603: Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone)

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1608 IP Telephone 1408 Digital Telephone

This telephone is best suited for the Everyday user.

Works on:	IP Office	Communication Manager
1608	✓	✓
1608-I	✓ (R5 and above)	✓
1408	✓	✓ (in future)

On the 1600 line, only the 1608-I version with improved display is available today.

The Avaya 1408/1608 supports 8 line appearances / feature keys. Each of the buttons includes dual LEDs (red, green) providing explicit status for the user. For a familiar look and feel, the 1408/1608 includes several fixed feature keys for common telephone tasks including conference, transfer, drop, hold, and mute. In addition, the 1408/1608 includes a high quality two-way speakerphone, and supports a broad portfolio of Avaya wired and wireless headsets through its integrated headset jack.

The 1408/1608 features a context sensitive user interface along with three softkeys and a four-way navigation cluster – ideal for scrolling through the local contacts list or call logs. The display on the 1408/1608 measures three lines by 24 characters and is backlit for easier viewing in all lighting conditions.

The newly introduced 1608-I is functionally equal to the 1608 but features a higher quality graphical display with white backlight, identical to the display of the digital 1416 telephone.

Common Features:

- Display/Housing:
  - Backlit display – 3 rows by 24 characters. Green backlight for 1608, white backlight for 1608-I, 1408.
  - Dual position flip stand.
- Fixed Buttons: 15 plus Four-way navigation cluster button.
  - Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button, Speaker button, Headset button.
  - Avaya Menu button (options and settings access), Telephony application button – to return to main telephone screen.
  - Hold button, Conference button, Transfer button, Drop button.

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- Contacts button, Call log button, Redial button, quick-access voicemail Message button.

- Programmable/Contextual buttons: 11.
  - 8 line appearance/feature key buttons – with dual LEDs (red, green) and paper labels (Printing tool available). Note: For typical installs, the first 3 buttons are reserved for standard call appearances; a minimum of 2 call appearance buttons is needed for standard operation.
  - Three contextual softkey buttons.
- Key Labels: Icons and text used on fixed feature keys.
- Features Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings, etc. On screen status indication for activated features like call forwarding.
- Speakerphone: High quality two-way handsfree speaker and microphone.
- Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.
- Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: Yes.
- Embedded Applications:
  - Centralized call log and contact application, fully transparent between 108/1608/1416/1616 phone, 9600 telephones and Avaya one-X™ Portal for IP Office.
  - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls).
  - Access to company directory and Visual Voice applications on IP Office.
- Upgradeable Firmware: Yes.
- Expansion: None.
- Color: Black.
- Mounting: Desk or wall mountable with optional wall mount adapter.

Requirements for 1408:

- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Requirements for IP interface (1608, 1608-I):

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) class 2 device or optional wall plug local power supply.
- Codecs/audio: G.711, G.729a/b, Dynamic Jitter buffer, Echo Cancellation, Comfort Noise, Automatic Gain Control.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN).
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Ethernet (10/100) line interface with a secondary 10/100 port for co-located laptop or PC with VLAN separation.
  - Well suited for everyday use, where typically smaller sized data files transferred (web, email file attachments) or less demanding data-transfer application/usage (thus less sensitive to data transfer completion times).
  - For highly demanding data traffic processing with large amounts of small packets we recommend the 1616 phone for full speed data throughput.
  - Support of optional Gigabit Ethernet Adapter.

Language Support:

- 1408/1608: Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone)

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1616 IP Telephone 1416 Digital Telephone

This telephone is best suited for the Receptionist/Assistant User.

Works on:	IP Office	Communication Manager
1616	✓	✓
1616-I	✓ (R6)	✓
1416	✓	✓ (in future)

On the 1600 line, only the 16016-I version with improved display is available today.

The Avaya 1416/1616 supports 16 line appearances/feature keys on the phone itself – and up to 3 x 32-button expansion modules, providing access to a total of 112 line/feature keys or speed dial buttons. This will satisfy the needs of even the most demanding secretary or receptionist. Each of the buttons features a dual LED (red, green) providing explicit status for the user. For a familiar look and feel, the 1416/1616 includes several fixed feature keys for common telephone tasks including conference, transfer, drop, hold, and mute. In addition, the 1416/1616 includes a high quality two-way speakerphone, and supports a broad portfolio of Avaya wired and wireless headsets through its integrated headset jack.

The 1416/1616 features a context sensitive user interface along with three softkeys and a four-way navigation cluster – ideal for scrolling through the local contacts list or call logs. The display on the 1416/1616 measures four lines by 24 characters and is adjustable for optimum viewing angle. Additional caller related information is displayed with active appearances for easier call handling. The display is backlit for easier viewing in all lighting conditions.

The newly introduced 1616-I is functionally equal to the 1616 but features a higher quality graphical display with white backlight, identical to the display of the digital 1416 telephone.

Common Features:

- Display/Housing:
  - Backlit display – 4 rows by 24 characters with adjustable display angle. Green backlight for 1616, white backlight for 1616-I, 1416.
  - Dual position flip stand.
- Fixed Buttons: 15 plus Four-way navigation cluster button.

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- Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button, Speaker button, Headset button.
- Avaya Menu button (options and settings access), Telephony application button – to return to main telephone screen.
- Hold button, Conference button, Transfer button, Drop button.
- Contacts button, Call log button, Redial button, Quick-access voicemail Message button.

- Programmable/Contextual buttons: 19.
  - 16 line appearance/feature key buttons – with dual LEDs (red, green) and paper labels (Printing tool available). Note: For typical installs, the first 3 buttons are reserved for standard call appearances; a minimum of 2 call appearance buttons is needed for standard operation.
  - Three contextual softkey buttons.
- Key Labels: Icons and text used on fixed feature keys.
- Feature Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings etc. On screen status indication for activated features like call forwarding.
- Speakerphone: High quality two-way handsfree speaker and microphone.
- Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.
- Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: Yes.
- Embedded Applications:
  - Centralized call log and contact application, fully transparent between 108/1608/1416/1616 phone, 9600 telephones and Avaya one-X™ Portal for IP Office.
  - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls).
  - Access to company directory and Visual Voice applications on IP Office.
- Upgradeable Firmware: Yes.
- Expansion: Module interface.
  - 1616: For up to 3 * BM32 32-button expansion modules (Note: An external power supply is needed for all installs with more than 1 button module).
  - 1416: for up to 3 * DBM32 32-button expansion modules (Note: An external power supply is needed in any case): Note: While the 1416 supports three DBM32, availability of the DBM32 will be slightly delayed compared to availability of 1416.
- Color: Black.
- Mounting: Desk or wall mountable with optional wall mount adapter.

Requirements for 1416:

- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Requirements for IP interface (1616 and 1616-I):

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) class 2 device or optional wall plug local power supply.
- Codecs/audio: G.711, G.729a/b, Dynamic Jitter buffer, Echo Cancellation, Comfort Noise, Automatic Gain Control.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN).
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Ethernet (10/100) line interface with a secondary 10/100 port for co-located laptop or PC with VLAN separation. Support of optional Gigabit Ethernet Adapter.

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Language Support:

- 1416/1616: Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone).

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BM32 Button Module for 1616 IP Phones DBM32 Button Module for 1416 Digital Phones

The DBM32/BM32 Button Module extends the number of feature buttons on your Avaya 1416 Digital/1616 IP Telephone. It provides 32 additional buttons all with dual LEDs (red, green) for easy visibility.

Up to 3 DBM32/BM32 button modules can be connected to a 1416/1616 phone. The total number of DBM32/BM32 button modules supported on one IP Office system is 32, subject to the total system limits.

Note that the DBM32 can only be used with the Digital 1416 while the BM32 is for use with the 1616 IP phone.

Typical uses are:

- to provide additional lines for calls bridged from one or more other extensions
- to provide abbreviated dialing buttons
- to provide buttons to access additional switch features like call forwarding

Please note that a external power supply is always needed for a DBM32 connected to a 1416 telephone and is needed when connecting more then one BM32 to a 1616 telephone.

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Other phone lines supported on IP Office

The following telephones are supported also on IP Office but not all of them are available as new telephones, e.g. the 5600 line of telephones has been discontinued.

Before offering a specific phone please check about availability. The information below about availability is accurate at the launch date of IP Office Released 7 but might change before this product description is updated.

Of course, the standard End of Sale process will apply before any product is discontinued- to make sure the channel has sufficient time to order required materials

First generation 9600 telephones:

The first generation of 9600 IP telephones can be an alternative to customers, especially if they are expanding installations that are using these telephones already. A wide choice of models, Color option, Gigabit Ethernet versions:,

- The 9620L/9620 C: a member of the Avaya one-X Deskphone Edition family, is specifically designed for the everyday telephone user—those who rely on multiple communications tools such as e-mail and IM, yet still require a high quality and intuitive telephone for voice communications. The 9620L IP Telephone features a high resolution 3.45 inch (9 cm) diagonal monochrome backlit display – while the 9620C features a color display for even better readability. The 9620L/C support up to 12 call appearances / administered feature keys - with three concurrent line appearances visible at any time.
- The Avaya 9650/9650C: The Avaya 9650 IP Telephone, a member of the Avaya one-X Deskphone Edition family, is specifically designed for the "navigator" telephone user. Navigators, such as building receptionists and executive admin staff, frequently manage calls for themselves and for groups of people. The 9650 features built-in button module functionality with one-touch access to bridged appearances, speed dials and feature keys. The Avaya 9650 IP Telephone delivers advanced communications capabilities, high definition audio, an integrated WML application interface, and comprehensive one-touch access. This is the ideal solution for receptionists, executive assistants, contact center agents as well as knowledge workers who need quick access to features and call appearances. The 9650 IP Telephone features a 3.8 inch (9.65 cm) diagonal ¼ VGA quality gray-scale pixel-based display with adjustable angle, while the 9650C enhances usability with a color display for even better readability. The 9650 supports up to 24 call appearances / administered feature keys.
- The Avaya 9630G/9640/9640G: a member of the Avaya one-X Deskphone Edition family, the 9640 IP Telephone with high resolution color display is specifically designed for the executive telephone user. The 9630G IP Telephone features a native Gigabit switch and a 3.8 inch (9.65 cm) diagonal monochrome backlit display - which has been enhanced with higher resolution (1/4 VGA) compared to other available monochrome telephones from Avaya. The 9640 IP Telephone features a 3.8 inch (9.65 cm) diagonal high resolution color backlit display. The 9640G model has the same functionality as the 9640, and adds native support for Gigabit. The 9630/9640 supports up to 24 call appearances/administered feature keys with six concurrent line appearances visible at any time.

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9620L, 9620C IP Telephones

Avaya 9620 IP Telephones are specifically designed for the everyday telephone user – those who typically rely on multiple communications tools such as e-mail and instant messaging (IM), yet still require a high quality and intuitive telephone for voice communications.

The Avaya one-X Deskphone family features an intuitive user interface which helps to make users proficient and confident in performing common telephone tasks such as setting up a conference call or completing a transfer.

With access to up to 12 lines and features through a simple-to-navigate interface, plus a clear view of 3 lines/features with LED indicators, the telephones are efficient to use while the small footprint consumes minimal space on a users desktop.

Works on:	IP Office	Communication Manager
9620L	✓	✓
9620C	✓	✓

Common Features:

- Display/Housing:
  - 7-line 3.45 inch backlight graphical display with 320*160 pixels. 9620L: grayscale; 9620C: color
  - Flip stand/dual position
- Fixed Buttons: 10 plus Four-way navigation cluster button.
  - Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED), Speaker button(LED), Headset button(LED).
  - Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen.
  - Contacts button, Call log button (LED), quick-access voicemail Message button with LED and corner message indicator.
- Programmable/Contextual buttons:
  - 4 contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.

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    - 12 virtual Appearance/Feature buttons, accessible through navigation cluster. Three features shown at a time with three LED indicators.
- Key Labels: Icons and text used on fixed feature keys.
- Features Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings etc. On screen status indication for activated features like call forwarding.
- Speakerphone: Full duplex speakerphone
- Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.
- Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: Yes.
- Embedded Applications:
    - Centralized call log and contact application, fully transparent between 9600 telephones, 1408/1416/1608/1616 telephones and one-X Portal for IP Office.
    - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls). Please note that the Voice-dial feature of the 9600 telephones on Communication Manager is not available on IP Office.
    - Access to company directory and Visual Voice applications on IP Office.
- External Applications: WML Application Interface
- Upgradeable Firmware: Yes.
- Expansion:
    - Adapter interface: 9620L none, 9620C, one expansion module interface for e.g. Gigabit Ethernet adapter.
    - USB adapter: 9620L none, 9620C: yes
- Color: Black.
- Mounting: Desk or wall mountable with optional wall mount adapter.
- Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone).
  Note: While it may be possible to download other languages onto the phone, this will not be applicable to any data sent by IP Office. If you switch the language to any other not listed above, the display will show a mixture of the new language and the user's default language on IP Office.

Requirements for IP Interface:

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or optional wall plug local power supply. 9620L: PoE Class 1, 9620C PoE Class 2
- Codecs/audio: G.711, G.729a/b, G.726, Dynamic Jitter buffer, Echo Cancellation, Comfort Noise, Automatic Gain Control.
  Ready for future support of G.722 wideband codec.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN).
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Ethernet (10/100) line interface with a secondary 10/100 port for co-located laptop or PC with VLAN separation.
- Support of optional Gigabit Ethernet Adapter: 9620L: external. 9620C integrated adapter using adapter interface.

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9630, 9640, 9640G Telephone

Smart, sleek, stylish and highly functional, the Avaya 9640 IP Telephone is designed for the Essential user, those for whom the telephone is essential in order to perform their jobs. Workers who conduct much of their business on the phone — moving from one call to the next. The Avaya 9630/9640 delivers advanced communications capabilities — high definition audio, a brilliant high resolution pixel based color display on the 9640, an integrated WML application interface, one-touch access to Avaya IP Office mobility or forwarding features — in a solution designed for those who are absolutely dependent on voice communications for their business operations.

With built in Gigabit Ethernet on the 9630G and 9640G, these telephones are also perfectly suited for customers with very high requirements for data network performance.

Works on:	IP Office	Communication Manager
9630G	✓	✓
9640	✓	✓
9640G	✓	✓

Common Features:

- Display/Housing:
  - 9-line 3.8 inch backlight graphical display with 320*240 pixels. 9650: grayscale; 9650C: color
  - Flip stand/dual position
- Fixed Buttons: 11 plus Four-way navigation cluster button.
  - Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED), Speaker button(LED), Headset button(LED).
  - Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen. Call Forward shortcut button
  - Contacts button, Call log button (LED), quick-access voicemail Message button with LED and corner message indicator.
- Programmable/Contextual buttons: 10.
  - 4 contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
  - 6 self-labeled Appearance/Feature buttons with LED for access to 24 administered Appearances/Features at the side of the display.
- Key Labels: Icons and text used on fixed feature keys.

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- Features Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings etc. On screen status indication for activated features like call forwarding.
- Speakerphone: Full duplex speakerphone
- Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.
- Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: Yes.
- Embedded Applications:
  - Centralized call log and contact application, fully transparent between 9600 telephones, 1408/1416/1608/1616 telephones and one-X Portal for IP Office.
  - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls). Please note that the Voice-dial feature of the 9600 telephones on Communication is not available on IP Office.
  - Access to company directory and Visual Voice applications on IP Office.
- External Applications: WML Application Interface
- Upgradeable Firmware: Yes.
- Expansion:
  - 2 Adapter interface:
  - USB 1.1 adapter
  - SBM24 Module interface for up to 3 x SBM 32-button expansion modules (Note: If more than one button module is used, Power Class of phone needs to be switched to Power Class 3)
- Color: Black.
- Mounting: Desk or wall mountable with optional wall mount adapter.
- Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone).
  Note: While it may be possible to download other languages onto the phone, this will not be applicable to any data sent by IP Office. If you switch the language to any not listed above, the display will show a mixed of the new language and the user's default language on IP Office.

Requirements for IP interface :

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or optional local power supply: Power Class 2 for all models including GiG-E versions.
- Codecs/audio: G.711, G.729a/b, G.726, Dynamic Jitter buffer, Echo Cancellation, Comfort Noise, Automatic Gain Control. Ready for future support of G.722 wideband codec.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN).
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports:
  - 9640: Ethernet (10/100) line interface with a secondary 10/100 port for co-located laptop or PC with VLAN separation.
  - 9630G/9640G: Gigabit Ethernet (10/100/1000) line interface with a secondary 10/100/1000 port for co-located laptop or PC with VLAN separation.
  - Support of optional integrated Gigabit Ethernet adapter using adapter interface.

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9650, 9650C IP Telephones

Smart, sleek, stylish and highly functional, the Avaya 9650 IP Telephone is designed for the Essential and Navigator user profiles who need one touch access to features and bridged appearances of team/Partner buttons. Building receptionists, executive admin staff are examples of Navigator users who answer incoming calls, transfer customers to other extensions and monitor several bridged appearances throughout the day. This is the ideal solution for receptionists, executive assistants, Contact Center Agents as well as Knowledge workers that need quick access to partner and team features.

Works on:	IP Office	Communication Manager
9650	✓	✓
9650C	✓	✓

Common Features:

- Display/Housing:
  - 9-line 3.8 inch backlight graphical display with 320*240 pixels. 9650: grayscale; 9650C: color
  - Flip stand/dual position
- Fixed Buttons: 10 plus Four-way navigation cluster button.
  - Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED), Speaker button(LED), Headset button(LED).
  - Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen.
  - Contacts button, Call log button (LED), quick-access voicemail Message button with LED and corner message indicator.
- Programmable/Contextual buttons: 15.
  - Four contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
  - 3 self-labeled Appearance/Feature buttons with LED for access to 24 administered Appearances/Features at the side of the display.

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    - 8 Additional self-labeled Appearance/Feature buttons with LED with direct access to 2*8 Appearances/Features.
- Key Labels: Icons and text used on fixed feature keys.
- Features Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings etc. On screen status indication for activated features like call forwarding.
- Speakerphone: Full duplex speakerphone
- Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.
- Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: Yes.
- Embedded Applications:
    - Centralized call log and contact application, fully transparent between 9600 telephones, 1408/1416/1608/1616 telephones and one-X Portal for IP Office.
    - Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls). Please note that the Voice-dial feature of the 9600 telephones on Communication is not available on IP Office.
    - Access to company directory and Visual Voice applications on IP Office.
- External Applications: WML Application Interface
- Upgradeable Firmware: Yes.
- Expansion:
    - 2 Adapter interface:
    - USB 1.1 adapter
    - SBM24 Module interface for up to 3 x SBM 32-button expansion modules (Note: If more than one button module is used, Power Class of phone needs to be switched to Power Class 3)
- Color: Black.
- Mounting: Desk or wall mountable with optional wall mount adapter.
- Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone).
  Note: While it may be possible to download other languages onto the phone, this will not be applicable to any data sent by IP Office. If you switch the language to any not listed above, the display will show a mixture of the new language and the user's default language on IP Office.

Requirements for IP interface:

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or optional local power supply: Power Class 2
- Codecs/audio: G.711, G.729a/b, G.726, Dynamic Jitter buffer, Echo Cancellation, Comfort Noise, Automatic Gain Control. Ready for future support of G.722 wideband codec.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN).
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Ethernet (10/100) line interface with a secondary 10/100 port for co-located laptop or PC with VLAN separation.
    - Support of optional integrated Gigabit Ethernet adapter using adapter interface.

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SBM24 Adapter Module

The SBM24 Expansion Module is an optional device that extends the number of call appearances and Feature buttons available on the telephone. The SBM24 Expansion Module is supported on the 9630, 9640 & 9650 IP telephones and provides 24 additional programmable buttons - to be configured as call appearances, bridged appearances, or feature keys. A maximum of three SBM24 modules are supported on a single phone.

- 24 programmable call appearance/feature keys.
- Two backlight LCD screens for button labels. Buttons automatically labeled from the system (no paper labels).
- Connects directly to the associated phone.
- No separate power supply needed. However, using more than one button module changes the power class of the phone to Class 3.

The total number of SBM24 button modules supported on one IP Office system is 42, subject to the total system limits.

A Button Module adapter (Ferrite adapter), is required when SBM24 button expansion module is used with the 9630G or 9640G IP telephones.

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Bluetooth Adapter

The Bluetooth Adapter for 9600 Series IP Telephones allows Bluetooth communications devices such as headsets to interoperate with 9600 Series telephones. The adapter allows highly mobile workers to use the same Bluetooth headset with their 9600 Series Deskphone as they use with their cell phone.

The Avaya ABT-35H Bluetooth headset when used with the Bluetooth Adapter supports a wireless range of up to 30 feet from the Deskphone and provides simple call control (answer, hang-up) directly from the headset.

For users while at their desk, the Bluetooth Adapter allows the freedom of hands free communication along with investment protection for those already utilizing Bluetooth headsets while on the go. It supports the Avaya ABT-35 as well as many other third party Bluetooth Headsets

The Bluetooth Adapter can be used with all 9600 telephones except the 9620L. It requires a separate "Wedge stand".

Gigabit Ethernet Adapter

The 9600 Gigabit Ethernet Adapter can be fully integrated into a wedge stand of the 9600 telephones and provides a transparent upgrade option if customers decide to install Gigabit Ethernet at a later date. It is fully integrated into the management interface of the 9600 telephones. It requires a separate "Wedge stand".

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5402, 5602 SW, 2402, 4602 SW Telephone

These telephones are best suited for the Walkup User.

Telephone	Works on IP Office	Works on IP Office and Communication Manager
5402	✓	✗
5602 SW	✓	✗
2402	✓*	✓
4602 SW	✓	✓

*Early 2402 telephones can make and receive call but the display will not function.
Note: The 5602 SW and 4602 SW telephones are supported by IP Office Release 6 but are no longer available for new sales.

Common Features:

- Display: 2 lines x 24 characters.
- Fixed Feature Buttons: 10 - Conference, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
- Programmable Feature Buttons:
  - DS telephones: 2 plus an additional 12 programmable feature keys can be accessed via the FEATURE key.
  - IP telephones: 2.
- Key Labels: Icons and text used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
- Speakerphone: Listen-only handsfree speaker (no microphone).
- Hearing Aid Compatible: Yes.
- Message Waiting Indicator: Yes. On the 2402 and 5402 this is can also be used as a ringing call alert indicator.
- Personalized Ring Patterns: Yes- 8 (via Conference button when idle).
- Headset Socket: No, this telephone does not support headset operation.
- Embedded Applications: None.
- Upgradeable Firmware: DS telephones - No. IP telephones - Yes.
- Expansion: None.
- Color: Multi-gray.
- Mounting: Desk or wall mountable.
- Adjustable Desk Stand: No.

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Requirements for 5402 and 2402:

- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Requirements for 5602 SW and 4602 SW:

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit (Avaya 1151 series).
- Codecs: G.711, G.729a/b.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
  - Auto-negotiation provided separately for each port.
  - 802.3 Flow Control.
  - Phone has priority over PC port at all times.

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5410, 5610 SW, 2410, 4610 SW Telephones

These telephones are best suited for the Receptionist/Assistant User.

Telephone	Works on IP Office	Works on IP Office and Communication Manager
5410	✓	✗
5610 SW	✓	✗
2410	✓	✓
4610 SW	✓	✓

Note that 4610, while supported on IP Office R6, is not longer available for new sales

Common Features:

- Display: 5 lines x 29 characters (168 x 80 pixel 4-grayscale).
- Fixed Feature Buttons: 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- Programmable Feature Buttons:
  - DS telephones: 12 - in 2 switchable display pages of 6 matching the 6 physical display buttons.
  - IP telephones: 24 - in 4 switchable display pages of 6 matching the 6 physical display buttons.
- Key Labels: Icons and text used on fixed feature keys.
- Speakerphone: Two-way handsfree speaker and microphone.
- Hearing Aid Compatible: Yes.
- Message Waiting Indicator: Yes - can also be used as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: Yes.
- Embedded Applications: Speed Dial List (48) and Call Log (Missed, Incoming, Outgoing). Also WAP WML browser supported on IP phone models.
- Upgradeable Firmware: Yes.
- Expansion: None.
- Color: Multi-gray.
- Mounting: Desk or wall mountable.
- Adjustable Desk Stand: Yes - Supplied with phone.

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Special Features for the 5410 and 2410:

- Messages Button: Dedicated button to collect voicemail.

Requirements for 5410 and 2410:

- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Requirements for 5610 and 4610:

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit (Avaya 1151 series).
- Codecs: G.711, G.729a/b.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
  - Auto-negotiation provided separately for each port.
  - 802.3 Flow Control.
  - Phone has priority over PC port at all times.

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5420, 5621, 2420, 4621, 4625 Telephones

These telephones are best suited for the Executive User and with the optional EU24 Button Module good for the Receptionist/Assistant User.

Telephone	Works on IP Office	Works on IP Office and Communication Manager
5420	✓	✗
5621 SW	✓	✗
2420	✓	✓
4621 SW	✓	✓
4625 SW	✓	✓

Note that 4621 and 4625, while supported on IP Office R7, but are not longer available for new sales.

Common Features:

- Display: 7 lines x 29 characters.
- Fixed Feature Buttons: 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- Programmable Feature Buttons:
  - DS telephones: 24 - arranged in 3 switchable display pages of 8 matching the 8 physical display buttons.
  - IP telephones: 24 - arranged in 2 switchable display pages of 12 matching the 12 physical display buttons.
- Key Labels: Icons and text used on fixed feature keys.
- Speakerphone: Two-way handsfree speaker and microphone.
- Hearing Aid Compatible: Yes.
- Message Waiting Indicator: Yes - can also be used as a ringing call alert indicator.
- Personalized Ring Patterns: Yes - 8.
- Headset Socket: Yes.
- Embedded Applications: Speed Dial List (104) and Call Log (Missed, Incoming, Outgoing). Also WAP WML browser supported on IP phone models.
- Upgradeable Firmware: Yes.
- Expansion: Supports the EU24 DSS expansion module (with additional Avaya 1151 power supply).
- Color: Multi-gray.
- Mounting: Desk or wall mountable.
- Adjustable Desk Stand: Yes - Supplied with phone.

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Special Features for the 5420 and 2420:

- Messages Button: Dedicated button to collect voicemail.

Requirements for 5420 and 2420:

- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Requirements for 5621 SW, 4621 SW, 4625 SW:

- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit (Avaya 1151 series).
- Codecs: G.711, G.729a/b.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- SNMP Support: Yes.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
  - Auto-negotiation provided separately for each port.
  - 802.3 Flow Control.
  - Phone has priority over PC port at all times.

Special Features for 5621 SW and 4621 SW:

- Display Backlight: The display has a backlight for improved contrast. Standby mode turns off backlight after time-out.

Special Features for 4625 SW:

- Color Backlight Display: The display is full color and has a backlight for improved contrast.

Note: While still supported, the 5620SW and 4620SW telephones are no longer available for purchase.

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EU24 and EU24 BL Expansion Modules

The EU24/EU24BL are phone expansion modules that work in association with a 5420, 5620/1, 2420, 4620/1, 4625 telephones. They provide an additional 24 programmable buttons with associated display label and status icons. Only one EU24 can be used per phone. The EU24BL has a backlight and is for use with the 4621 and 5621 only.

The EU24/EU24BL supports an additional 24 Call Appearance/Feature buttons, by displaying a column of 12 buttons at a time, with a dotted line separating the two columns.

Common Features

- 24 Programmable call appearance/feature keys.
- Automatically labeled from the system (no paper labels).
- Connects directly to the associated phone.
- Requires an Avaya 1151 series power supply, even for IP telephones already using Power over Ethernet (PoE).
- IP Office supports a maximum of eight EU24/EU24 BL's on each IP Office system, subject to total system limits.

Telephone	EU24	EU24BL
2402/5402	✗	✗
2410/5410	✗	✗
2420/5420	✓	✗
4601/5601	✗	✗
4602/5602	✗	✗
4610/5610	✗	✗
4620/5620	✓	✗
4621/5621	✓	✓
4625	✓	✓

Note that EU24 BL, while supported on IP Office R6, is not longer available for new sales.

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1100 and 1200 IP Phones

The 1100 IP telephones are inherited from the Norstar and BCM product line. These telephones are supported on the BCM platform using “Unistim” protocol.
On IP Office these telephones are supported using the SIP protocol. SIP software can be downloaded onto those telephones when they get connected to IP Office. When used with IP Office, the telephones need an “Avaya IP phone” license like any other Avaya IP phone. No other license is needed (e.g. with IP Office no separate IP-phone license is required.

The following telephones can be migrated:

1100 series: 1120E and 1140E plus Key Expansion module
1200 series: 1220 and 1230 plus both Key Expansion modules.

Please note that for new installs Avaya recommends the 9600 line of telephones like the 9608, 9621 or 9641. They telephones offer the full access to all innovative IP Office features like Visual Voice-Mail, IP Office Feature menu or centralized call-logs or personnel directories.

1100 and 1200 IP Phones: overview

Users can enjoy the next-generation features of VoIP, plus all of the carrier-class reliability and ease of use of traditional telephones. These telephones offer four superior desktop models, along with two innovative software-based solutions that bring VoIP to a user’s computer or PDA. The IP telephonesinclude:

- IP Phone 1100 Series – The IP Phone 1120E and 1140E are IP color display telephones. They offer a fully pixel-based, high-resolution, backlit display. They are optimized for Web-centric applications such as phone support under XAS/G-XAS. They also provide USB support for peripherals such as a keyboards or mouse.
- Key Expansion Module for the IP Phone 1100 Series – The Expansion Module for IP Phone 1100 Series is a hardware accessory that will provide up to 18 additional line / programmable feature keys per module. The module is supported on the IP Phone 1120E and the IP Phone 1140E.
- IP Phone 1200 Series – The IP telephones 1220, and 1230 are BCMs family of IP Phone tailored for the SMB market and now also supported on IP Office. They provide a rich set of capabilities at a price competitive for the SMB market.
- Key Expansion Module for the IP Phone 1200 Series – The Expansion Module for IP Phone 1200 Series is a hardware accessory that is available in two options. A 12 key LCD based model or a 18 key based LED module. The modules are supported on the IP Phone 1220 and the IP Phone 1230.
- Please note that the following telephones, available with BCM, are not supported on IP Office: 1210, 1110.

IP Phones Benefits

- Support connectivity to heritage Nortel VoIP enabled platform, including BCM and Communication Server 1000 using Unistim as well as IP Office using the SIP protocol
- Transnetwork support increases employee productivity by providing users and network managers with a common set of telephones across the entire network
- Internal voice/data switch prioritizes voice traffic to ensure high-quality speech and reduces costs by conserving wiring closet ports and eliminating the need for separate cable drops to the desktop
- Automatic firmware upgrade ensures top performance and streamlines maintenance, reducing the need for site visits
- DHCPenabled IP telephones simplify network administration by providing centralized, automated IP address management

Multiple Platform Support
These IP telephones are supported by heritage Nortel communication systems, including BCM, and Communication Server 1000 but now are available also for IP Office. This industry-leading platform interoperability facilitates growth and offers seamless migrations across customer premises and carrier-based solutions. All IP telephones offer a rich suite of business features designed to meet current and emerging user requirements and international icon-based characters. These features, in addition to multilanguage support, make the IP telephones ideal for worldwide use.
Integrated Switched Ethernet Connection
Users can connect their desktop Internet telephone and their PC to the network on a single port. The IP telephones feature a built-in 10/100 Base T Layer 2 switch that splits the network Category 5 cable into separate feeds, one to the phone itself, while providing an additional RJ-45 port to connect a user’s PC. Giving fixed, hardware-based priority to the phone voice port, the internal Ethernet switch ensures that high-quality voice service is always available.
Reliable LAN Power Options
To ensure continuous phone service in mission-critical environments, the IP telephones can receive their power over the network cabling. The Avaya family of Ethernet Switchs (BES 50, ERS2500, ERS4500) delivers power over the unused pairs of standard Category 5 UTP cables, eliminating the need for the telephones to be connected to a power supply at the desktop.
Power is supplied on an as-needed basis, thanks to the sophisticated, software based load and fault-sensing

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algorithm used by the Avaya Ethernet Switches. Desktop, rackmount and stackable options are available which provide a cost-effective way to centralize power to the IP telephones. This approach delivers carrier-grade reliability by enabling redundant power resources located at the wiring closets to provide emergency backup power to these IP telephones located across the network.

Dynamic Host Configuration Protocol
Easy to set up and configure, the IP telephones deliver an innovative solution that enables users to connect anywhere on the network without intervention by a network administrator, enabling management staff to focus on more complex, mission-critical responsibilities.
Whether users relocate their phone down the hall or across the globe, the service comes up in the new location exactly as if they were sitting in their own office, even though it might be thousands of miles away.
Prioritizing Network Traffic
Now companies can make their network's priorities mirror those of their business. By installing the Avaya Ethernet Switch (BES50 or ERS), a business' network becomes an intelligent partner that uses Layer 2/3 packet classification to prioritize business-critical traffic, helping them get the most out of their existing infrastructure. Latency-sensitive applications and mission-critical users will receive platinum-level service, while less urgent traffic is allocated bandwidth on a lower-priority basis. Prioritization is especially important for businesses that need to support mission-critical IP applications, including VoIP, but do not want to incur higher costs by overprovisioning the network to ensure bandwidth availability.
Universal Features
With over 100 years of experience in creating telephones designed to improve employee productivity, reduce operational expenses and improve customer service, Avaya is the industry leader in telephony solutions. All of the IP telephones offer the following features:

- Intuitive navigation cluster provides fast menu, sublist and call log scrolling, as well as one-touch dialing and quick access to system features
- Message waiting/visual ring indicator offers visual notification of incoming calls and messages
- Voice compression optimizes bandwidth and audio quality requirements
- Audio control center enables users to toggle quickly between the handset or headset and the speakerphone without audio interference
- Volume bar provides fingertip control of audio and ringer volume settings and LEDs clearly display handset/ headset/speakerphone/mute settings
- Local tone generation conserves valuable network bandwidth
- Dynamic IP addressing with a standard DHCP server offers a flexible, simplified solution for handling adds, moves and changes, reducing management costs
- User-friendly design supports the full range of potential users, including disabled users who require hearing aids.

SIP Protocol: Feature overview on IP Office with 1100/1200 phones

The telephones are supported on IP Office using the SIP protocol, therefore the user interface can be slightly different then when used on BCM with "Unistim!".
Note that these telephones are only supported on IP Office 500/500 V2 hardware.

They are supporting a large set of features including:

- Standard phone features like hold, transfer, conference (IP Office based)
- Multiple call appearances (note that currently bridged appearances and trunk-line appearances are not supported)
- Message waiting support
- Busy Lamp keys with speed-dial, status indication and pickup of ringing calls
- Application support. The telephones can be used in combination with applications like "one-X Portal" for added comfort, allowing e.g. click and dial functionality.
- Self labelling Feature keys: This allows access to selected IP Office features using a single key-press. If additional parameters are needed, the user is guided using a dialog interface of feature request plus softkeys for e.g. confirmation
- "Special" feature key. The supported features are also available pressing a "feature" (soft-)key plus the appropriate feature code. The feature codes are identical to the features codes from BCM, therefore making transition easy and painless.
- Ease of installation: The telephones can be managed from IP Office manager and are easily installed. IP Office will serve as DHCP-server and can provide confguration files, software updates etc. To install the telephones, only IP Office manager is needed.
- When migrating telephones from BCM to IP Office, when IP Office DHCP server is used the system will take care of installing/upgrading the telephones to the correct software fo IP Office. This makes migration easy and quick.
-

Feature Table: (next to the below features that are accessible through the "Feature key" and can be administered to a button, the telephones can also access IP Office features by dialling the IP Office Feature Access code- plus parameter if needed - for the desired feature.)

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Feature	Details
F*0 – Button query (also allows DN query via I/C button)	supported
F*1, F*2, F*3 – Self-admin button programming	supported
F*6 – Self-admin ring type	function supported on the phone but not using the feature code
F*7 – Self-admin contrast	function supported on the phone but not using the feature code
F*80 – Self-admin ring volume	function supported on the phone but not using the feature code
F0 – Speeddial	supported
F3 – Conference	supported
F4 – Call Forward, F#4 cancel	supported
F5 – Last Number Redial	supported
F60 – Page (no F61-F63)	supported
F66 – Dial Voice Call	supported
F70 - Transfer	function supported on the phone but not using the feature code
F74 – Call Park, F#74 Retrieve	supported
F75 – Call Pickup Group	supported
F76 – Call Pickup Extn	supported
F85 – Do Not Disturb, F#85 cancel	supported
F981 – Enter Voice Mail System (same as F65) AUDIO ONLY	supported
F812 – Call Log	function supported on the phone but not using the feature code

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IP Phone 1120E

IP Phone 1140E

The AVAYA IP telephones 1120E and 1140E offer an IP desktop solution that is optimized for Web-centric applications such as phone support, and other multimedia options. They feature Gigabit Ethernet, USB connectivity, and an ergonomic design and cable management that make them easy to set up and maintain. The 1120E and 1140E IP telephones are supported on IP Office R6.1 and higher

Phone 1120E

The IP Phone 1120E provides the following features:

- A multi-line phone with four programmable line/feature key appearances
- High-resolution backlit graphical pixel-based display
- Adjustable five-position tilt-stand (-5 to + 55°), plus vertical design for smaller desktop footprint
- Fourteen fixed keys and four soft keys
- 10/100/1000 Base-T with integrated PC & LAN Ports
- XAS/G-XAS supports data applications and Web browsing
- USB port for enhanced navigation and productivity
- Expansion Module/Console Port
- Visual Alerting and Feature Status Indicator Lamps
- Integrated headset port for optional wired Headset
- Active Ethernet link LED indication
- Secure Tools Menu facilitates access to preferences & administration
- 802.3af PoE (Class 3) or local power options
- Desk and Wall Mountable
- Color: Graphite with Silver Metallic Bezel

IP Phone 1140

The IP Phone 1140E provides the following features:

- Multi-line phone, with up to 12 programmable line/feature keys
- High-resolution backlit graphical pixel-based display
- Adjustable five-position tilt-stand (-5 to +55°)
- Fourteen fixed keys and four soft keys
- Integrated Bluetooth radio for office and home-based mobility
- 10/100/1000 Base-T with integrated PC & LAN Ports

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- XAS/G-XAS supports data applications and web browsing
- USB port for enhanced navigation and productivity
- Visual Alerting and Feature Status Indicator Lamps
- Expansion Module/Console Port
- Integrated headset port for optional wired Headset
- Active Ethernet link LED indication
- Secure Tools Menu facilitates preferences & administration
- Global Character Sets
- 802.3af PoE or local power option
- Color: Graphite with Silver Metallic Bezel

IP Key Expansion Module for IP Phone 1100 Series

IP Phone 1100 Series Expansion Module

The IP Phone 1100 Series includes a Key Expansion Module that provides up to 18 additional line/programmable feature keys per module. The module is supported on the IP Phone 1120E and the IP Phone 1140E.

Nortel IP Phones 1220, and 1230

Nortel IP telephones 1220, and 1230

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The IP Phone 1200 series, a group of IP telephones Launched with BCM are now also available with IP Office. They are tailored for the SMB market. The 1200 series provides a range of different telephones designed to address the needs of the SMB market with a broad feature set at competitive prices.
IP Phone 1220 is an intermediate IP phone targeted at most SMB employees. The IP Phone 1230 is a premium IP phone for managers and advanced/power users.

IP Phone 1220
The IP Phone 1220 provides the following features:

- A five-line display with 25 characters per line
- A fixed stand with two positions, plus vertical design for smaller desktop footprint
- Eight fixed keys, with three having LED status
- Hands free, Mute, Headset, Volume Controls, Hold, Goodbye, Applications.
- Six feature keys
- Four context-sensitive soft keys
- Four line/feature keys
- 10/100 Base-T with integrated PC Port
- Active Ethernet link LED indication
- Integrated headset port for optional wired Headset
- Compatible with optional Key Expansion Modules
- Visual ringing alerter / Message Waiting Indicator
- 802.3af PoE (Class 2) or optional AC local power options
- Wall Mountable

IP Phone 1230
The IP Phone 1230 provides the following features:

- A nine-line display with 25 characters per line
- A fixed stand with two positions, plus vertical design for smaller desktop footprint
- Eight fixed keys, with three having LED status
- Handsfree, Mute, Headset, Volume Controls, Hold, Goodbye, Applications.
- Six feature keys
- Four context-sensitive soft keys
- 10 line/feature keys
- 10/100 Base-T with integrated PC Port
- Active Ethernet link LED indication
- Integrated headset port for optional wired Headset
- Compatible with optional Key Expansion Modules
- Visual ringing alerter / Message Waiting Indicator
- 802.3af PoE (Class 2) or optional AC loc

IP Key Expansion Modules for IP Phone 1200 Series

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LCD IP Key Expansion Module

LED IP Key Expansion Module

The IP Phone 1200 Series includes two Key Expansion Modules to provide flexibility for adding more keys to the IP Phone 1220 and IP Phone 1230. The extra keys can be used for additional line appearances or to add often-used features for quicker access. The keys can also be programmed as auto-diallers for one-touch calling, and will provide busy lamp field indication for users on the BCM system. This allows these telephones to better serve the needs of attendant and receptionist functions, and increase productivity for any user. Key features include:

- LCD Expansion Module for the IP Phone 1200 Series offers 12 keys per module, cascadable to 7 LCD KEMs on an IP Phone 1220 or 1230
- LED / Paper Label Module for the IP Phone 1200 Series offers 18 keys per module, with LED indicators for line status and an easy-to-use paper labeling system, cascadable to 2 LED Paper-Label KEMs on an IP Phone 1220 or 1230.

Business Series Terminals (Digital phones)

Business Series Terminals offer a feature-rich portfolio with enhanced capabilities that provide telephony solutions for a broad landscape of users, from high-volume call positions and executives to low-intensive users and small workgroups.
The Business Series Terminals were originally positioned for deployment on two system platforms from the Nortel Heritage, Norstar and BCM, providing both investment protection and a migration path between either system. The Business Series Terminals offer full integration with Norstar and BCM features, as well as integration with basic and advanced applications such as Messaging, Intelligent Contact Center, Computer Telephony Integration (CTI) and integrated voice and data solutions.
The telephones are now available also on IP Office and offer customers primarily another migration option from Norstar or BCM to IP Office.
Please note that those terminals do not work on all digital ports of IP Office but require a specific 8-port expansion card or 16/30 port expansion module: For details refer to the IP Office hardware segment:

Recommendation: The BST digital phone support on IP Office is mainly intended for migration from Norstar to BCM. For new installs, it is possible to use the BST telephones. For new installs Avaya recommends using the 1400 or 9500 digital sets in order to get access to all advanced features o IP Office like Visual Voice Mail, Feature menus or self labelling keys (9500 telephones only)

Product Overview:

While the Business Series Terminals boast the industry leadership and strengths of the BCM and Norstar telephone portfolio, the portfolio also delivers value-added features. These telephones are now also supported on the IP Office platform. Feature highlights include:

- Tilt Display – provides clearer viewing of information or message prompts on the LCD in different lighting environments.
- Message waiting indication/Visual ringing lamp – alerts the user of incoming messages or that their phone is ringing when they are on another call.
- Headset interface – is driven from the Digital Terminal Interface Chip (DTIC). Volume control for the headset is also provided. Operation of the headset is mutually exclusive, with Handsfree operation. When a headset is connected, all operations normally associated with handsfree operation affect the headset. This includes on-hook dialing, volume control while active and muting.
- Handsfree interface – is programmed through the administration function and is supported by a microphone and loudspeaker.

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- External ringer interface – receives alerting signals that are routed to the external ringer jack as well as to the speaker in the telephone. This alerting signal can be amplified and connected to external speakers to provide an auxiliary ringer function for the telephone. The external speaker is connected with a two-wire modular telephone cord to pins 3 and 4 of the external ringer jack.

The portfolio also offers tilt display, new aesthetics, a streamlined footprint, new labeling strategy, an audio control center with a headset button and more.

Sets are available in the following colors:

- Platinum
- Charcoal (not T7000)

On IP Office, the following features are available using the “feature key” and the Feature access code inherited from Norstar/BCM: Other IP Office features are also available by dialing the IP Office Feature Access code:

F*0 – Button query (also allows DN query via I/C button)

F*1, F*2, F*3 – Self-admin button programming

F*6 – Self-admin ring type

F*7 – Self-admin contrast

F*80 – Self-admin ring volume

F*82 – Dial mode selection (Enbloc/Pre-dial [editable], or Standard/Overlap dial)

F*84 – Set call log options (All, Missed, Answer, Outgoing) (we've concluded that this adds no value)

F0 – Speeddial

F2 – Ring Again, F#2 cancel

F3 – Conference

F4 – Call Forward, F#4 cancel

F5 – Last Number Redial

F60 – Page (no F61-F63)

F65 – Retrieve Messages - Enters Voice Mail System on IPO (Basic Embedded Voicemail)

F66 – Dial Voice Call

F69 – Dial Priority Call

F70 - Transfer

F74 – Call Park, F#74 Retrieve

F75 – Call Pickup Group

F76 – Call Pickup Extn

F85 – Do Not Disturb, F#85 cancel

F981 – Enter Voice Mail System (same as F65) AUDIO ONLY

F812 – Call Log

F802 – Group Listen, F#802 cancel

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Overview of T7000

The T7000 telephone is an entry-level digital set for basic low-use office environments. It has four programmable feature keys. It does not have display buttons (soft keys), and it does not support headset or Handsfree.
The T7000 is used primarily in the EMEA (not available in North America).
The T7000 supports the following features:

- External ringer interface
- Message waiting indicator/visual ringing lamp
- Dial-up feature access

Button Matrix

The T7000 buttons are as follows:

- Twelve dial pad buttons
- Volume control (rocker type)
- Hold button
- Release button
- Feature button
- Four programmable button

Overview of T7100

The T7100 telephone is part of the Business Series Terminals portfolio. It has one programmable button and a 1 x 16 character alphanumeric display to provide call progress information. There are no display buttons (soft keys) on the T7100 and it does not support headset or Handsfree.
T7100 supports the following features:

- External ringer interface
- Message waiting indicator/visual ringing lamp
- LCD with tilt Display – The one-line by 16-character display on the T7100 telephone provides call progress information. Each character is generated from a matrix of 5 x 7 LCD dots under control of a built-in controller chip on the module.

Button Matrix

The T7100 buttons are as follows:

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- Twelve dial pad buttons
- Volume control (rocker type)
- Hold button
- Release button
- Feature button
- One programmable button

Loop Limits

- Maximum loop length – 305 m (1000 ft) or 0.5 mm (24AWG) wire
- Maximum loop length – 790 m (2600 ft) with SAPS option
- Bridge taps – not permitted; loading coils – not permitted.

Overview of T7208

The T7208 telephone is part of the Business Series Terminals portfolio. It has eight fully programmable buttons, each with its own LCD indicator and a 1 x 16 character alphanumeric display to provide call progress information. There are no display buttons (soft keys) on the T7208.

The T7208 supports the following features:

- External ringer interface
- Headset interface
- Handsfree interface
- Message waiting indicator/visual ringing lamp
- LCD with tilt Display – The one-line by 16-character display on the T7208 telephone provides call progress information. Each character is generated from a matrix of 5 x 7 LCD dots under control of a built-in controller chip on the module.

Button Matrix

The T7208 buttons are as follows:

- Twelve dial pad buttons
- Volume control (rocker type)
- Hold button
- Release button
- Feature button
- Eight programmable buttons with indicators.

Loop Limits

- Maximum loop length – 305m (1000 ft) of 0.5 mm (24 AWG) wire
- Maximum loop length – 790 m (2600 ft) with SAPS option
- Bridge taps – not permitted; loading coils – not permitted.

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Overview of T7316E

The T7316E telephone is part of the Business Series Terminals portfolio. It has 24 programmable buttons and a 2 x 16 character alphanumeric display that is menu driven and supported by three context-sensitive soft keys. The T7316E provides access to 24 memory buttons, 16 of which include multi-segment icons for fast and precise decision-making.
The T7316E supports the following features:

- Multi-line w/multi-segment call appearance icons
- 16 programmable buttons for lines/features/autodials
- 8 additional memory buttons for features/autodials
- Expandable by adding T24 KIM
- Integrated BLF/DSS
- Fixed buttons: feature, hold, release
- Two-line adjustable display w/soft keys
- Built-in speakerphone
- Message Waiting Indication
- Time and date displayed
- Built-in headset
- Audio control center with mute, hands free, headset and volume bar
- Default features based on profiles
- Default line and hunt group assignment
- Retractable quick reference card
- Desk or wall mount
- Hearing aid compatible
- Desktop Assistant label application compatible
- ITU dial pad
- Colors: charcoal and platinum
- English and French keycaps (accessory)
- Extra length 2.7m handset cord (accessory)
- Shoulder rest (accessory)

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Overview of BST Central Answering Position (T7316E + T24)

The BST Central Answering Position is an expandable desktop telephone that allows administrative assistants and emergency contact centers the ability to centralize and efficiently distribute calls. By attaching the T24 Key Indicator Module (KIM) directly to the T7316E, office administrators and emergency contact centers can transform the pace and efficiency of their operations while maintaining an exceptional level of customer service.
T24 Product features:

- BCM and Norstar compatible
- Two versions of the T24 are available (Enhanced for line appearances and Ordinary for features and autodialers)
- Multi-line with multi-segment call appearance icons
- 24 programmable buttons for lines/features/autodials
- Busy Lamp Field/Direct Station Select support
- Multiple appearances of hunt groups
- Multiple line appearances
- 4 T24's per T7316E without power supply
- 9 T24's per T7316E with power supply
- Desk or wall mount
- Colors: charcoal and platinum

The following table highlights configuration rules:

	Enhanced KIM (use KIM for multiple appearances of target lines)			Ordinary KIM (use KIM for BLF/DSS)		
Max # sets per system	Max # sets	Max # EKIM on each set	EKIM (max #)	Max # sets	Max # OKIM on each set	Max OKIM
96	12	4	48	84	9	756

Avaya Audio Conferencing Unit

Description

Designed in partnership with Polycom, the Avaya Audio Conferencing Unit offers superior teleconferencing by using three microphones to provide 360 degrees of voice coverage. The Conferencing Unit is a full duplex handsfree unit. This feature allows voice to be heard and picked up at the same time, providing faster response time and eliminating conversation "collisions" and losses. The unit has a Feature Key that allows access to many

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of the same features found on the Business Series Terminals The Conferencing units connects to the same digital ports of IP Office that are also supporting BST terminals.

IP Office supports a number of different button modules to provide telephones with additional programmable buttons. The limits stated below are those applicable for the IP Office 4.2 2Q 2009 maintenance release and higher.

- BM32: *(1616)*
  Up to 32 BM32s supported (32 buttons each, up to 1024 buttons).
- DSS4450: *(4412D+, 4424D+)*
  Up to 8 DSS4450 modules per system (maximum of 2 per DS expansion module, including the IP406v2) (50 buttons each, max 400 buttons).
- DBM32: *(1416)*
  Up to 32 DBM32s supported (32 buttons each, up to 1024 buttons).
- EU24/EU24BL: *(4620, 4621, 5420, 5620, 5621/4620, 4621, 5620, 5621)*
  Up to 8 EU24s per system (24 buttons each, max 196 buttons).
- SBM24: *(9630G, 9640, 9640G, 9650, 9650C)*
  Up to 42 SBM24s per system (24 buttons each, up to 1008 buttons).
- BM12: *(9508. 9608,9611 9641)*

  Up to 42 SBM24s per system (24 buttons each, up to 1008 buttons).
- BM12: *(9608,9611 9641)*

  Up to 42 SBM24s per system (24 buttons each, up to 1008 buttons).
- LCD IP key Expansion module for 1200 telephones *(1220,1230)*

  Up to 82 LCD-IP KEMs per system (12 buttons each, up to 1008 buttons).
- LED IP key Expansion module for 1200 telephones *(1220,1230)*

  Up to 56 LED-IP KEMs per system (18 buttons each, up to 1008 buttons).
- LCD IP key Expansion module for 1100 telephones *(1220,1230)*

  Up to 56 LCD-IP KEMs 24s per system (12 buttons each, up to 1008 buttons).
- T3 Series: *(All T3 telephones)*
  Up to 30 T3 DSS modules supported. (36 buttons each, up to 1080 buttons). T3 DSS buttons are not included in the overall system limit for additional button module buttons.
- XM24: *(6416+, 6424+)*
  Up to 10 XM24 units per system (maximum of 2 per DS expansion module, including the IP406v2) with a maximum of 1 per 6416+/6424+ phone (24 buttons each giving a maximum of 240 XM24 buttons).
- T24 KIM (Key Indicator Module)☹ T7316)
  Up to 42 T24 KIMs per system (24 buttons each, up to 1008 buttons).

The maximum number of additional button module buttons per system is 1024 (regardless of if the buttons are configured for use or not). Only exception is use of T3 DSS only where, if no other button modules are used, a total of 1080 buttons is supported:

- 12 x BM32 + 5 x EU24 + 20 x SBM24 is allowed (984 buttons in total).
- 15 x T3DSS + 15 x BM32 is allowed (1020 buttons in total).
- but NOT 8 x BM32 + 10 x EU24 (because it exceeds the maximum number of 8 EU24s, even though total buttons < 1024).
- and NOT 14 x BM32 + 5 x EU24 + 20 x SBM24 (total 1048 buttons, this exceeds the maximum 1024 buttons per system).

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T3 Telephones

The T3 range of digital Upn and IP telephones provide European style with context sensitive displays and are available in select European countries only.While the digital version is currently available, the IP version is no longer sold new. We recommend to use the 9600 line IP telephones instead.

T3 Compact

This telephone is best suited for the Walkup User. It is available in Europe only.

Note that the T3 IP compact is no longer available for new sales

Common Features:

- Display: 1 line with 24 characters alphanumerical plus one line of icons
- Fixed Feature Buttons: 3 keys with printed text labels and 2 for Audio Volume control
- Programmable Feature Buttons: 3 keys with indicators and printed text labels, 2 keys with printed text labels
- Speakerphone: Full duplex handsfree speaker and microphone.
- Hearing Aid Compatible: Through optional handset
- Message Waiting and call log Indicator: Yes
- Personalized Ring Patterns: Yes, 8 ring patterns
- Embedded Applications: Navigation Cursor Control, Call signaling via LED and/or ringer; Alphanumeric entry via dialing keypad.
- Color: Graphite gray or polar white.
- Mounting: Desk or optional wall mountable.
- Adjustable Desk Stand: No

Features for T3 Upn only:

- Upgradeable Firmware: No.
- Optional Add-Ons: Up to 3 DSS Modules. T3 Headset link for wired headsets.
- Headset Socket: No. Optional headset add-on available.
- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Features for T3 IP only:

- Upgradeable Firmware: Yes
- Headset Socket: Yes
- Optional Add-Ons: No
- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit.
- Codecs: G.711, G.729a/b.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- SNMP Support: No.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Two port full-duplex 10/100 BaseT Ethernet switch.
  - Auto-negotiation provided separately for each port.

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- 802.3 Flow Control.

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T3 Classic

This telephone is best suited for the Everyday User. It is available in Europe only.

Note that the T3 IP Classic is no longer available for new sales

Common Features:

- Display: 4 lines x 26 characters
- Fixed Feature Buttons: 5 preprogrammed keys with printed text labels and 2 for Audio Volume control
- Programmable Feature Buttons: 6 preprogrammed keys with indicators and printed text labels, 4 programmable keys with printed text labels
- Speakerphone: Two-way handsfree speaker and microphone.
- Hearing Aid Compatible: Through optional handset
- Message Waiting and call log Indicator: Yes
- Personalized Ring Patterns: Yes, 8 ring patterns.
- Headset Socket: No. Optional headset add-on available.
- Embedded Applications: Navigation Cursor Control, Call signaling via LED and/or ringer; Alpha entry via dialing keypad.
- Optional Add-Ons: Up to 3 DSS Modules. T3 Headset link for wired headsets.
- Color: Graphite gray or polar white.
- Mounting: Desk
- Adjustable Desk Stand: Display adjustable

Features for T3 Upn:

- Upgradeable Firmware: No.
- Optional Add-Ons: up to 3 DSS Modules
- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Features for T3 IP:

- Upgradeable Firmware: Yes.
- Optional Add-Ons: up to 3 DSS Modules with AEI/Headsetlink,
- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit.
- Codecs: G.711, G.729a/b.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- SNMP Support: no.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Two port full-duplex 10/100 BaseT Ethernet switch.
  - Auto-negotiation provided separately for each port.
  - 802.3 Flow Control.

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T3 Comfort

This telephone is best suited for the Executive as well as the Receptionist/Assistant User. It is available in Europe only.

Note that the T3 IP Comfort is no longer available for new sales

Common Features:

- Display: 17 lines x 40 characters, Integrated keyboard
- Fixed Feature Buttons: 5 preprogrammed keys with printed text labels and 2 for Audio Volume control
- Programmable Feature Buttons: 6 preprogrammed keys with indicators and printed text labels, 6 preprogrammed keys with printed text labels, 10 user programmable keys with associated display labels.
- Speakerphone: Two-way handsfree speaker and microphone.
- Hearing Aid Compatible: Through optional handset
- Message Waiting and call log Indicator: Yes
- Personalized Ring Patterns: Yes, 8 ring patterns.
- Headset Socket: No. Optional headset add-on available.
- Embedded Applications: Navigation Cursor Control, Call signaling via LED and/or ringer
- Optional Add-Ons: Up to 3 DSS Modules. T3 Headset link for wired headsets.
- Color: Graphite gray or polar white.
- Mounting: Desk
- Adjustable Desk Stand: Display adjustable

Features for T3 Upn:

- Upgradeable Firmware: No.
- Optional Add-Ons: up to 3 DSS Modules
- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Features for T3 IP:

- Upgradeable Firmware: Yes.
- Optional Add-Ons: up to 3 DSS Modules with AEI/Headsetlink,
- Power Supply: IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit.
- Codecs: G.711, G.729a/b.
- QoS Options: UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- SNMP Support: No.
- IP Address Assignment: Static or dynamic IP address assignment.
- Ethernet Ports: Two port full-duplex 10/100 BaseT Ethernet switch.
  - Auto-negotiation provided separately for each port.
  - 802.3 Flow Control.

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T3 DSS Expansion Modules

The T3 DSS Module is a phone expansion module that is compatible with all T3 Upn and T3 IP Telephones except the T3 IP Compact. Each module provides an additional 36 programmable buttons with associated printed text labels and indicators, and can be programmed for lines, groups or speed dial numbers. 3 DSS Modules can be added to each T3 phone. Power is provided by T3 Upn telephones, but an external power adapter is needed for each DSS module when used on T3 IP telephones.

IP Office 406, IP412 and IP500 support a maximum of 30 T3 DSS modules per control unit. Please refer to System Button Module Support for the total system limitations of button modules on IP Office.

Wireless and Remote Phone Solutions

Avaya IP Office Wireless Solutions include analog, digital and IP-based WiFi wireless telephones. These are solutions employees can use every day to work more effectively and be more responsive to customers - all while increasing revenues and keeping communication costs firmly under control. Also, Avaya IP Office Mobility Solutions integrate seamlessly with IP Office, enhancing each customer's investment. IP Office's in-building Mobility Solutions improve communication with staff that, because of the function they perform, are mobile within the workplace. Using wireless technology, such individuals may be instantly contactable, with many obvious benefits;

- The wireless telephone is carried in the pocket, so users are not tied to the desk in order to remain in contact.
- Users may be contacted instantly to ensure fast, accurate decision making and immediate response to problems through planned radio coverage with no blind spots

Avaya Wireless and Remote Solutions

IP Office supports the following wireless solutions:

- DECT R4, is the mainstream IP based DECT solution, offered in the EMEA and NA regions and in selected APAC countries. With the new Compact base-station, it is also the first choice for small installs with smaller channel/capacity need.
- IP-DECT in the EMEA and NA regions and in selected APAC countries is also supported on IP Office. Note that the IP-DECT solution will be eventually completely replaced by DECT R4.
- Digital Wireless North American market.
- Avaya VoIP WiFi Solution offered worldwide in selected countries including North America, EU countries and selected others.

IP Office supports the following VPN remote phone solutions:

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- VPN phone functionality on all supported 9600 telephones. VPN functionality is a integral part of the 9600 software and does neither require a separate software load nor does it require a license.

Wireless - DECT R4

The Avaya DECT R4 solution delivers the productivity-boosting benefits of IP and wireless communications across multiple offices in a convenient, lightweight handset. It provides businesses with a highly functional wireless solution with the ability to scale to support large numbers of users. This system also supports users in different offices connected via a WAN. The Avaya DECT R4 solution radio fixed part or IP base station (IPBS) connects to the IP Office using an IP protocol based on H.323.

Avaya's DECT R4 solution features all advantages of a full blown DECT solution for the enterprise market: Cost effective high wireless voice quality in a frequency band exclusively reserved for DECT that is secure, easy to deploy and enhance.

What is new with R7

With IP Office R6 the cost efficient 4-channel base-station "compact" was introduced, offering a more economical solution for small installs of e.g. 3 base-stations/10 handsets where 4-radion channels are sufficient:

IP Office Release 7.0 extends the features of DECT R4, including integrated administration and simplified install. This not only exposes more IP Office features to the DECT solution, but additionally reduces the TCO for both business partner and customer.

Highlights include:

- Simplified install - 4 steps on the master base station, nothing on the slave; rest by handset prompts or IP Office Manager
- Simplified GUI mode on the base station displays options that are applicable to IP Office only
- All further administration via IP Office Manager, e.g. Adds. Moves and changes..
- Integrated handset in-call and out of call feature menu giving access to many features without programming the handset: Login/out, forwarding (including number), park/hold, conference etc
- Idle line display of user name/number including status indicator Do Not Disturb, parked/held calls, twinned, group etc.
- Additional display of calling number during alerting, using two lines of display
- Two new, special purpose handsets:
  - One ruggedized handset for use in e.g., industrial environments offering superb protection from liquids, dust etc.: 3740:
  - A special "intrinsically safe" handset for usage in areas with high risk of explosions, like coal-mines, refineries, gas stations etc. where e.g. a minor internal spark in the handset could otherwise ignite a explosion.: 3749
- DECT R4 is now available in additional regions, especially in the CALA theatre. Type approval has been achieved for Mexico, Brazil and Argentina. Please note that due to regulation, separate power supplies are required and not all powering/charging options may be available. Please refer to the DECT R4 product description on support.avava.com

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System Overview

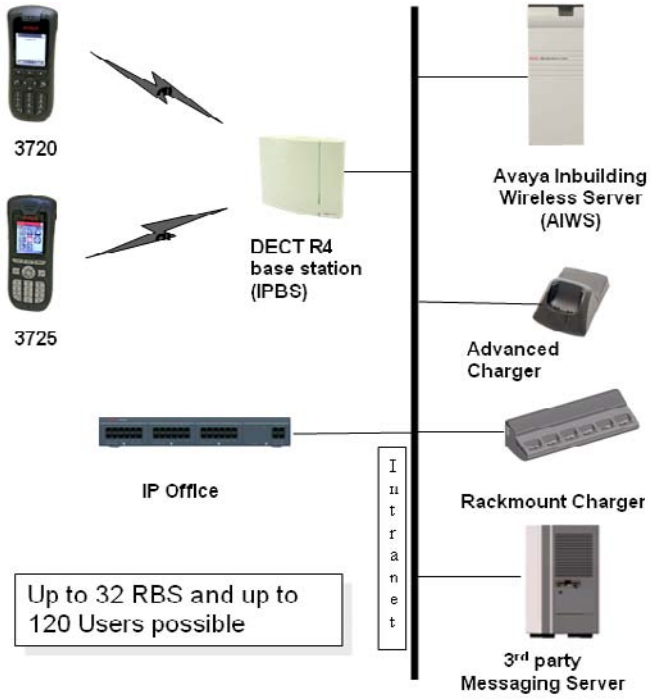
The complete DECT R4 solution, including the new R6 components, consists of the following:

- Two wireless handsets (3720 and 3725) plus a line of accessories.
- Two 8-channel radio base stations (with internal and external antennas) with IP interface for usage with IP Office.
- A 4-channel Compact base-station for small installs, up to 5 can be used per system.
- An appliance server for centralized functions like corporate directory and internal phonebook access, simple text messaging, integration of messaging and other external applications, centralized configuration and maintenance etc.

As the solution contains a handset with liquid protection and BlueTooth headset interface and an appliance server for attaching messaging applications it is especially well suited for verticals like healthcare and retail.

All handsets and radio base stations support the DECT frequency bands in EMEA, APAC, NAR and CALA with the same hardware and firmware. (Please check type approval for availability in the different regions)

The Avaya DECT R4 solution on IP Office supports up to 120 handsets and 32 base stations. Each base station can be powered over the LAN using the Power over Ethernet (PoE) standard. Each base station can be also powered by an external power supply. There are two versions of the base station (IPBS) available:, a standard base station with fixed antennas and an enhanced base station with a variety of external antennas (External antenna not available in the US). Both base-stations can be installed in an outdoor housing to provide an "external" base-station



In all approved regions, this system supports the 3720 and 3725 handsets.

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IP DECT Radio Base Station (RBS)

The DECT R4 Radio Base Stations are used with IP Office only. It connects directly to a LAN switch using Ethernet cabling. The DECT R4 Radio Base Stations can either be powered by Power-over-Ethernet or by using external power supplies that are available as accessories.

Each standard DECT R4 Radio Base Station can handle up to eight concurrent calls while each Compact Radio Base Station can handle up to 4 concurrent calls. A special protocol called on-air synchronization ensures that active calls are seamlessly handed over from one DECT R4 Radio Base Station to the next one if a user with an active call roams through a building.

To make sure that the on-air protocol works and that the whole building is covered by the ranges of the DECT R4 Radio Base Stations a site survey is needed to determine the number and placement of the DECT R4 Radio Base stations within the building. Avaya offers these site surveys as a service offer by AGS as well as measurement kits for business partners who want to do these site surveys on their own.

The on-air synchronization as well as the access to the phonebook of IP Office is done by a special software (master server) that can run in parallel to the standard DECT R4 Radio Base Station software on the same hardware.

DECT R4 Radio Base Station with internal antennas

DECT R4 Radio Base Station with external antennas

Each base station has the following features:

- Two variants of the standard Radio Base Station: With Internal antenna and with External Antenna due to regulation, in the US only the base-station with internal antenna may be used.
- There are several antennas available for the base-station with external antenna: Standard, Directional Single, Directional Dual, Omnidirectional.
- One variant of the Compact Radio Base Station with internal antennas. The compact base station has the same look and feel as the base station with internal antenna.
- Outdoor housing available for both versions
- 8 Simultaneous voice and up to 12 signaling channels for the standard Radio Base Stations.
- 4 Simultaneous voice and up to 12 signaling channels for the compact Radio Base Stations.
- Codec G.711, G723, G.729a/b for base station IP trunk connection.
- Power Consumption
- Power over Ethernet or local power supply supported
- Power over Ethernet, IEEE 802.3af, class 2, typical 4W, maximum 5W consumption.
- Dimension: 200 (w) x 165 (d) x 56 (h) mm including mounting bracket
- Weight RBS: 450g.

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DECT Feature Overview

Given the degree of integration available to wireless users with DECT, there are a variety of means by which calls can be routed to wireless handsets:

- Handover
  While in motion, the handset performs continuous measurements to determine which DECT R4 base station has the strongest signal. The one that can be best received is defined as the active Base station. To prevent the handset from rapidly switching back and forth between two base stations that are equally well received, threshold values are used. Handover between base stations occurs seamlessly whether a call is active or not.
- DECT Networking
  An DECT R4 telephone can travel from one office to another which is connected over a wide area network (WAN) link and make and take calls. In this scenario the main DECT R4 controller remains at one "headquarters" location, with slave base stations in up to 31 other offices.
- DDI/DID
  Since each wireless handset is an extension on the IP Office system calls may be routed directly using a DDI/DID number.
- Transfer
  Calls may be transferred to DECT extensions by operators or other extension users and DECT extension users may transfer callers to any other extension user.
- Hunt group compatibility
  Wireless handsets may be programmed as members of groups and answer calls in the same manner as any other extension within that group.
- Group working
  Wireless handsets may be programmed as members of groups and attract calls in the same manner as any other extension within that group. DECT handsets must NOT be configured into collective groups.
- Divert destination
  Users may initiate any or all diverts from an Avaya desk phone to a wireless handset.
- Twinning
  Twinning allows calls to a user main extension number to alert at both that extension and a secondary extension. Though not restricted to DECT, this feature is aimed primarily at users who have both a desk phone and a wireless extension. Calls from the secondary twinned extension are presented as if from the users main extension. Presentation of call waiting and busy is based on whether either of the twinned extensions is in use.
- DECT R4 Capacities:

Feature	DECT R4
Maximum Handsets	120
Maximum Base Stations if Master Server is on a standard RBS	32
Maximum Base Stations if Master Server is on a Compact RBS	5
Total Base Stations/Compact Base Stations	32
Total number of Compact Base Stations	5
Maximum simultaneous calls	100*

  *May be limited by the available VCM voice compression channels for calls to non-IP destinations. DECT R4 on IP Office does not support Redundancy option.

- Licensing and Redundancy
  Other then IP-Dect, Dect R4 does not require any licensing by itself (next to the licensing for the appropriate VCM channels in IP Office). Dect R4 on IP Office does not support Redundancy option

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DECT R4 Inbuilding Wireless Server (AIWS)

The Avaya Inbuilding Wireless Server enhances the functionality of the DECT R4 with additional functionality such as IP Office and LDAP directory access, centralized management and software update as well as handset to handset SMS plus basic web messaging.

Avaya In-Building Wireless Server (AIWS) is a server for handling wireless services and messaging. It runs on a solid state embedded Server (AIWS Hardware). It comes pre-installed, with the appropriate features activated on that server.

For installation and maintenance, this server is managed by a PC using Windows Internet Explore (6.0 or above) and Sun's Java Runtime Environment.

Several variants of the server are available. All Servers come fully installed and preloaded with licenses so no extra installation is needed. There is no upgrade available from e.g. Basic to standard.

	Basic	Basic+	Standard	OAP
NTP Server *(IP Office can also act as NTP Server)*	X	X	X	X
Access To Central Phonebook	X	X	X	X
Access To Corporate Directories *(LDAP on CM /TFTP on IPO)* *(from release 6, central IP Office phonebook is also supported directly on the Master RBS)*	X (only IPO)	X (only IPO)	X (IPO & LDAP)	
SMS Handset to Handset	X	X	X	X
Basic Web Messaging	X	X	X	X
Netpage Web Messaging			X	
Software Download Over-The-Air		X (up to 32 handsets)	X (up to 120 handsets)	
Software Download Via Advanced/Rack Charger		X (up to 32 handsets)	X (up to 120 handsets)	
Centralized Handset Configuration Over-The-Air		X (up to 32 handsets)	X (up to 120 handsets)	
Centralized Handset Configuration Via Advanced/Rack Charger		X (up to 32 handsets)	X (up to 120 handsets)	
Virtual SIM Card		X (up to 32 handsets)	X (up to 120 handsets)	
AIWS as Protocol Converter				X (only OAP)

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DECT R4 Accessories

In addition to the standard chargers, a number of high functionality chargers exist:

- Basic Charger
  - Charges one DECT phone (3720 and 3725)
  - Charge time: 4 hours.

- Advanced Charger
  - Charges one DECT phone (3720 and 3725)
  - Supports easy replacement of telephones
  - Connected to configuration tool PDM (Portable Device Manager) via USB or Ethernet
  - Charge time: 4 hours.

- Rackmount Charger
  - Same functionality as the advanced charger but for up to 6 DECT telephones (3720 and 3725)
  - Charge time: 4 hours.

- Multiple Battery Charger
  - Charges 6 batteries (3725).

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- Portable Device Manager
  Software to be used in combination with Advanced or Rackmount Charger in order to Manage the 3720 and 3725 phone:
  - Requirements: Windows XP® Professional, Service Pack 2 (SP2) or Windows Vista® Business operating system, Sun Java Runtime Environment (JRE) 6 or later, Acrobat Reader 5 or higher.

- Other accessories
  - Handset Basic Belt Clip
  - Handset Swivel Belt Clip
  - Handset Leather Case (3720 and 3725)
  - Handset Security Chain

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IP-DECT, while fully supported on IP Office R7, is no longer available for new sales or upgrades.

The IP DECT solution delivers the productivity-boosting benefits of IP and wireless communications across multiple offices in a convenient, lightweight handset. It provides businesses with a highly functional wireless solution with the ability to scale to support large numbers of users. This system also supports users in different offices connected via a WAN. The Avaya IP DECT solution radio fixed part (RFP) or base station connects to the IP Office using an IP protocol based on H.323.

The Avaya IP DECT solution supports up to 120 handsets and 32 base stations. Each base station can be powered over the LAN using the Power over Ethernet (PoE) standard. Each indoor base station can also optionally be connected to main power via an external power adaptor. Each outdoor base station can only be powered using PoE - no individual power supplies are available to power the outdoor IP DECT base station.

In EMEA and APAC this system supports the 3701 and 3711 handsets.

In North America, only the 3711 handset is supported.

Note: The regulatory requirements for the radio part (base station and Handset) are slightly different in the US and Canada compared to EMEA and APAC. Therefore, while providing the same functionality, the hardware is different in these two regions.

Each Base station has the following features:

- 8 simultaneous Voice and up to 12 Signaling Channels.
- Codec G.711, G.723, G.729 for base station IP trunk connection.
- Handover

  While in motion, the handset performs continuous measurements to determine which IP DECT base station has the strongest signal. The one that can be best received is defined as the active Base station. To prevent the handset from rapidly switching back and forth between two base stations that are equally well received, threshold values are used. Handover between base stations occurs seamlessly whether a call is active or not.
- DECT Networking

  An IP DECT telephone can travel from one office to another which is connected over a wide area network (WAN) link and make and take calls. In this scenario the main IP DECT controller remains at one "headquarters" location.

Given the degree of integration available to wireless users with DECT, there are a variety of means by which calls can be routed to wireless handsets:

- DDI/DID
  Since each wireless handset is an extension on the IP Office system calls may be routed directly using a DDI/DID number.
- Transfer
  Calls may be transferred to DECT extensions by operators or other extension users and DECT extension users may transfer callers to any other extension user.
- Hunt group compatibility
  Wireless handsets may be programmed as members of groups and answer calls in the same manner as any other extension within that group.
- Group working
  Wireless handsets may be programmed as members of groups and attract calls in the same manner as any other extension within that group. DECT handsets must NOT be configured into collective groups.
- Divert destination
  Users may initiate any or all diverts from an Avaya desk phone to a wireless handset.
- Twinning
  Twinning allows calls to a user main extension number to alert at both that extension and a secondary extension. Though not restricted to DECT, this feature is aimed primarily at users who have both a desk phone and a wireless extension. Calls from the secondary twinned extension are presented as if from the users main extension. Presentation of call waiting and busy is based on whether either of the twinned extensions is in use. In North America this functionality became available in Release 4.0.7.

IP DECT Capacities

Maximum handsets	120
Maximum base-stations	32
Total base-stations/repeaters	32
Maximum simultaneous calls	100*

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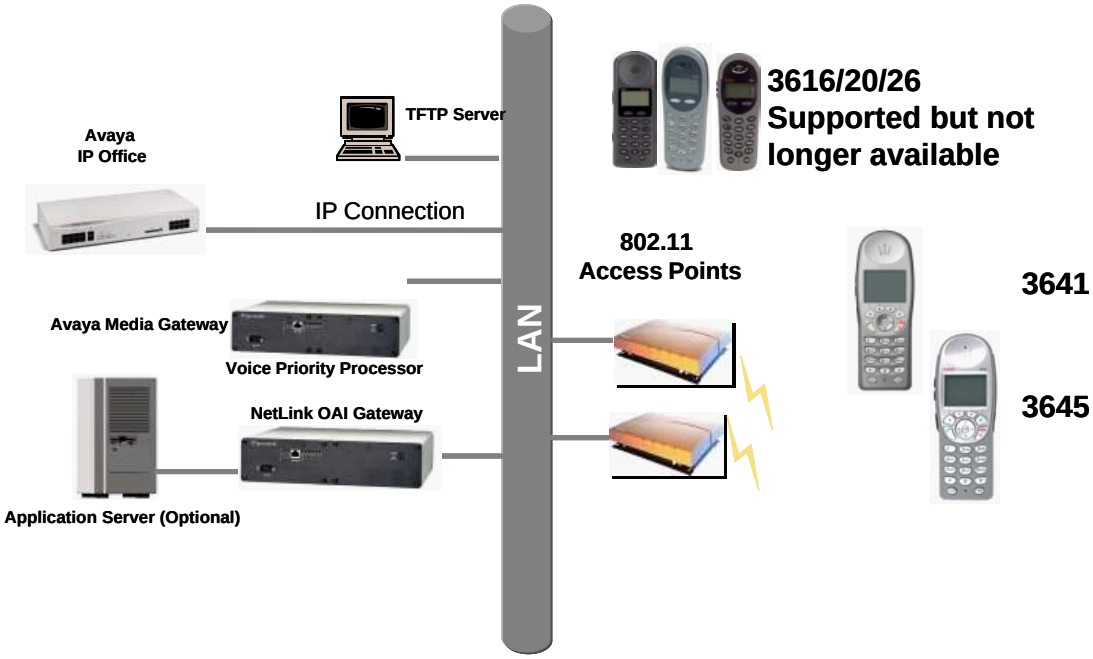
*May be limited by the available VCM voice compression channels for calls to non-IP destinations.

Wireless - WiFi (802.11)

The Avaya IP Wireless solution offers an advanced Voice over IP (VoIP) client for wireless networks. This solution allows SME's to take advantage of the cost savings and simplified management of a converged voice and data infrastructure.



The 3641 and 3645 telephones provide an improved user-interface, a new lightweight design and a radio that supports several WiFi standards (802.11a/b/g). With these handsets customers have an increased choice to fit their needs and infrastructure.

Based on global standards for wireless LAN's, the Avaya IP Wireless Telephone Solution simplifies network infrastructure by enabling voice traffic to be carried along with data traffic over the same wireless network. 3616, 3620 and 3626 telephones are supported but no longer available from Avaya for direct sequence 802.11b WiFi networks; the 3641 and 3645 will also work in 802.11a and 802.11g networks. These telephones are also field upgradeable through external TFTP clients (not included), so telephones can be updated with new protocols, features, and capabilities as they become available.

Users can have a choice of an executive or rugged workplace telephone and all the productivity benefits of their desk telephone in this next generation of wireless telephone solutions.

Wireless IP Terminals

Users can have a choice of 2 WiFi telephones to meet their in building mobility needs:

- Both support the 802.11 a/b/g standards and up to 12 lines or selected features. Both of these handsets are resistant to dust and spraying water and therefore also suitable for harsh environments. They also offer office-quality speaker-phone functionality.
  - Avaya 3641 supports a broad range of enterprise applications and is ideally suited for general office, financial or hospitality industries. This compact handset offers a high-resolution backlight graphic display a new, improved user-interface and design and a lightweight form factor.
  - Avaya 3645 is a slightly larger version that in addition supports "push-to-talk" functionality for broadcast communication between employees. Due to its rubberized sized grips and the larger ear cup it is especially well suited in noisy and industrial environments.
  - Access to 2*6 lines or selected features through the "line" key and the "FCN" key (no individual labels are supported for the administered functions on the function keys, status information only for call appearances but not for line appearances).

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Standards based Quality of Service support

Since Summer 2010 IP Office R6 Maintenance release a new software was made available for the 3641 and 3645 handsets.

This software allows to use open standard Quality of Service mechanisms based on WMM standards.

Wi-Fi Multimedia (WMM) is a specification and certification program created by the Wi-Fi Alliance, based upon the Enhanced Distributed Channel Access (EDCA) mechanism defined in the 802.11e standard. Avaya handsets support two QoS methods; the proprietary SpectraLink Voice Priority (SVP) developed to meet a gap which existed before 802.11e ratification and the standards-based Wi-Fi Alliance Wireless Multi-Media (WMM). SVP or WMM-enabled APs and an AVPP are required for all Avaya wireless telephone installations, even if the WLAN is being used only for voice. Without a method of prioritization for voice packets, the lack of a controlled delivery method will result in poor audio quality, reduced capacity, poor handoffs and/or poor battery life even with only voice devices on the network. In a mixed configuration using the AVPP and WMM refer to the Configuration and Deployment Guides for Polycom VIEW Certified and Field Verified WLANs for specific configuration requirements.

Basic WMM implementation in access points does not include Unscheduled-Automatic Power Save Delivery (U-APSD) and Call Admission Control (CAC), which are also defined in the 802.11e standard. Therefore, the AVPP is required to ensure the timing and delivery of Avaya voice packets, which is critical as the user hands off between APs and for improved battery life. The AVPP also provides call admission control and allows the number of calls per AP to be configured.

Please also refer to the Deployment Guide for Avaya-Polycom VoWiFi Telephones available on support.avaya.com

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Avaya Voice Priority Processors

The Avaya Voice Priority Processor (AVPP) is an Ethernet LAN appliance that works with access points to provide Quality of Service (QoS) on the wireless LAN. All packets to and from the wireless telephonespass through the AVPP and are encapsulated for prioritization as they are routed to and from IP Office. AVPP is fully compliant with the IEEE 802.11a/b/g standards.

AVPP is required for QoS because the current IEEE 802.11a/b/g wireless LAN standards provide only limited mechanism for differentiating audio packets from data packets. It also delivers quality of service by limiting the number of telephones that are connected to one access point in order to avoid quality problems. In addition AVPP ensures that the phone can run in energy-efficient mode when not in use. The following AVPPs are available to meet customer needs:

- AVPP100: Serves 80 calls simultaneously.
- AVPP020: Serves 20 powered-on handsets.
- AVPP010: Serves 10 powered-on handsets.

Wireless Access Points

When using the Avaya Wireless IP solution, customers can utilize wireless access points from various vendors. The list of compatible wireless access points is large and constantly growing. Please visit http://www.polycom.com/support/voice/wi-fi/view_certified.html and select "WLAN Compatibility List" for the latest information.

Benefits

- Supports 802.11 a/b/g standards for Wi-Fi networks converging voice and data over a single network.
- Seamless integration with IP Office.
- Excellent voice quality on converged wireless networks.
- Lightweight, durable handsets specifically designed for workplace use.
- Improved display, battery life, processor power all with lower costs.
- Increased range of AVPP's to address the needs of diverse construct sizes.
- Multitude of accessories are available:
  - Dual Charger (full charge accomplished in approximately one and a half hours).
  - Quick Charger (full charge accomplished in approximately one and a half hours).
  - Single, Dual, and Quad Chargers for the 3641 and 3645 telephones.
  - Belt Clip.
  - Nylon Pouch.
  - Carrying case with Lanyard.
  - Hands Free Pouch.
  - Noise canceling headset.
  - Over the ear headset.

Avaya IP Wireless Telephony Solution (AWTS) Open Application Interface (OAI) Gateway

The AWTS Open Application Interface (OAI) Gateway enables third- party software applications to communicate with the Avaya IP Wireless Telephones. This serves as a two-way messaging device. Many companies provide applications that interface to your in-house paging systems, email, and client-server messaging. Other vendors with complementary systems such as nurse call, telemetry, alarm, and control system manufacturers are currently developing applications to interface with the Avaya IP Wireless Telephone solution.

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3616 Wireless Telephone

While no longer available, this WiFi telephone is still supported on IP Office Release 6.

The Avaya 3616 IP Wireless Telephone is a WiFi (802.11b) telephone that runs using H.323.

The 3616 supports the following features:

- Lightweight innovative design .
- Simple to use.
- 802.11b standard-compatible.
- Radio Frequency 2.4000 – 2.835 GHz (SMI).
- Transmission type Direct Sequence Spread Spectrum (DSSS).
- FCC certification Part 15.247.
- Management of telephones via DHCP and TFTP.
- Voice encoding G711.
- Transmit Power 100mw peak, <10mW average.
- Wired Equivalent Privacy (WEP), 40bit and 128 bit.
- 2x16 character alphanumeric, plus status indicators.
- 4 hours talk time and 80 hours standby.

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3626 Ruggedized Wireless Telephone

While no longer available, this WiFi telephone is still supported on IP Office Release 6.

The Avaya 3626 Wireless Telephone is a WiFi standard (802.11b) telephone that runs using H.323.

The 3626 supports all of the features of 3616 with the following differences:

- Designed for industrial environments.
- Ruggedized durable design.
- Push-to-talk (walkie-talkie) feature for broadcast communications between employees.

Note: 3626 supports both R1.0 and R2.0 firmware on the set itself. However, as of R3.1 of IP Office, only 3626 phone R1.0 firmware is supported.

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3641 Ruggedized Wireless Telephone

The Avaya 3641 Wireless Telephone is a WiFi standard (802.11a/b/g) telephone that runs using H.323.

The 3641 supports the following features:

- Slim lightweight design with large display.
- Backlight display with Icons.
- Simple to use with improved user interface.
- Navigation and soft keys for simple access to frequently used operations.
- Office-quality speakerphone for handsfree operation.
- 802.11a/b/g standard-compatible.
- Radio Frequency 2.4000 GHz (b/g) or 5.8 GHz (a).
- FCC certification Part 15.247.
- Management of telephones via DHCP and TFTP.
- Voice encoding G711, G.729a.
- Wired Equivalent Privacy (WEP), 40bit and 128 bit and 802.11i (PSK) for secure communication.
- Lithium Ion Battery pack with up to 8 hours talk time and 160 hours standby.
- IP-53 Design (Liquid/dust protection).
- MIL 810F Design (Shock protection).
- Clips, cases, lanyard.

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3645 Ruggedized Wireless Telephone

The Avaya 3645 Wireless Telephone is a WiFi standard (802.11a/b/g) telephone that runs using H.323.

The 3645 supports all of the features of 3641 166 with the following additions:

- Push-to-talk (PTT) functionality for workgroup communication
- Enlarged earpiece for operation in noisy environments
- Rubberized grips for improved ergonomics and durability

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3720 DECT

This telephone is fully supported on the Avaya DECT R4 system.

However, it can also be used on an Avaya IP DECT system in "GAP-mode" with functionality limited to standard calling functionality, DECT roaming and handover. Access to IP Office central directory as well as time of day update is not available in GAP-mode.

Standard DECT handset:

- Black and white display with Backlight and Icons
- Loudspeaker / hands free
- Graphical user interface
- Four way navigation key
- 6 built-in UI languages
- Standby 180 hours , Talk Time 16 hours (under optimal conditions)

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3725 DECT

This telephone is fully supported on the Avaya DECT R4 system

However, it can also be used on an Avaya IP DECT system in 'GAP-mode' with functionality limited to standard calling functionality, DECT roaming and handover. Access to IP Office central directory as well as time of day update is not available in GAP-mode.

Standard DECT handset:

- Color Display with Backlight and Icons
- Loudspeaker / hands free
- Build in Bluetooth 2.0, supporting the "headset" profile. While others standard compliant headsets should work, Avaya has only tested a selected number. See list of Headsets behind the feature table.
- Graphical user interface
- Four way navigation key
- 19 built-in UI languages. However, in some languages menu items may appear in the English language.
- Standby 180h, Talk Time 16h (under optimal conditions) Bluetooth headset interface (Bluetooth 2.0, handsfree profile)
- Liquid and dust protected (IP 44)
- Multi-functional button (alarm call, answer call, etc.)
- Text message support (requires AIWS, 30 messages sent/received storable, message length 160 characters)

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Avaya 3720 and 3725 DECT Handsets Feature Overview

	Telephone	3720 DECT Telephone	3725 DECT Telephone
Keys:	Keypad	yes	yes
	Hook on/off buttons	yes	yes
	Softkeys	3	3
	Navigation cluster	4-way	5-way
	Volume buttons		yes
	Mute button		yes
	Alarm button		yes
Display:	Display, illuminated	112 x 115 pixels, Black and white	128 x 160 pixels, 64K colors
	Battery indicator	yes	
	Signal strength indicator	yes	
	Time and date display	yes	
Audio:	Handsfree	yes, 2-way speakerphone	
	Headset interface 2.5 mm	yes	
	Bluetooth 2.0 Headset profile	no	yes
	Supported Ring tones	14	14
	Ring tone volume	8 levels	
	Vibration	yes	
	Ringing options	Ring, Vibration, visual	
	Usage profiles	supported	
	Keyes beep	yes, selectable	
	Volume control	8 levels	8 levels
Features	Message LED (Red-green)	yes	yes
	Call log (Local)	25 entries	25 entries
	Local Directory/Contacts	250	250
	Access to IP Office Directory	Possible[1]	Possible[1]
	Support of Text Messaging	no	Optional[1]
	Phone lock	pin protected	pin protected
	Keyes lock	yes	yes
	Programmable "in Call Feature" menu	10 functions	10 functions
	Programmable "Call Services" Menu	10 functions	10 functions
	GAP compliant	yes	yes
Hardware:	Battery	Lithium Polymer	Lithium-Ion
	Standby/Talk time - No Bluetooth	160/16 hours	120/20 hours
	Standby/Talk time - with Bluetooth	n.a.	120/13 hours
	Charge time	4 hours	4 hours
	Dust/Liquid Protection level		IP 44
	Ruggedness	IEC 68-2-32, procedure 1, dropped 12 times from 1 metre	
	Size	133 × 53 × 24	134 × 53 × 26 mm

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	Telephone	3720 DECT Telephone	3725 DECT Telephone
	Weight	115 g (battery and clip included	130 g (battery and clip included)
	Belt clip options	none, standard, swivel	
Localization:	Supported languages	English,	Danish
		German	Dutch
		Spanish	English
		French	Finnish
		Russian[2]	French
		Downloadable language (one of 14[2])	German
			Italian
			Norwegian
			Portuguese
			Spanish
			Swedish
			Polish[2]
			Greek[2]
			Czech[2]
			Hungarian[2]
			Brasilian[2]
			Slovakian[2]
			Turkish[2]
			Russian[2]

1. Requires AIWS Server for integration with IP Office
2. Language supported on phone, IP Office menu items might appear in English.

The new DECT R4 handsets will work with the legacy IP DECT system. Nevertheless not all features available with the 3701 and 3711 handsets will be available on the DECT R4 handsets as well. An (incomplete) list of working features is:

- Subscription with PARK.
- Basic Call.
- Display Messages (with some limitations, e.g. truncation at end of line).
- Handover.
- Roaming.
- R-Key Handling (Enquiry call, Conference, ...).
- Feature Access Codes.
- Distinguish between internal/external calls.
- DTMF during call.
- Local call log (calling party numbers are transmitted).
- Local time and date.

Features already known to be not working with DECT R4 handsets and the legacy IP DECT system are

- Access of external directory (e.g. LDAP).
- Get time and date from the system.

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- WML (WAP access).
- SMS/Messaging (not a feature of the legacy IP DECT system).

Generally it is highly recommended to do a sample installation with one handset first before rolling out a larger set of DECT R4 handsets on a legacy IP DECT system

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This telephone is fully supported on the Avaya DECT R4 system.

However, it can also be used on an Avaya IP DECT system in "GAP-mode" with functionality limited to standard calling

- Black and white display with Backlight and Icons
- Loudspeaker / hands free
- Graphical user interface
- Four way navigation key
- 19 built-in UI languages. However, in some languages menu items may appear in the English language.
- Standby 150h, Talk Time 18h (under optimal conditions)
- Ruggedness: Shockproof certification IEC 60068-2-32, procedure 1 from 2 meters
- Operating temperature: -10°C to +55°C
- Liquid and dust protected (IP 65)
- Multi-functional button (alarm call, answer call, etc.)
- Text message support (requires AIWS, 30 messages sent/received storable, message length 160 characters)

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3749 Intrinsically save DECT handsets

This telephone is fully supported on the Avaya DECT R4 system.

However, it can also be used on an Avaya IP DECT system in "GAP-mode" with functionality limited to standard calling functionality, DECT roaming and handover. Access to IP Office central directory as well as time of day update is not available in

Color display with Backlight and Icons

- Loudspeaker / hands free
- Build in Bluetooth 2.0, supporting the "headset" profile. While others standard compliant headsets should work, Avaya has only tested a selected number. See list of Headsets behind the feature table.
- Graphical user interface
- Four way navigation key
- 19 built-in UI languages. However, in some languages menu items may appear in the English language.
- Standby 150h, Talk Time 18h (under optimal conditions)
- Ruggedness: Shockproof certification IEC 60068-2-32, procedure 1 from 2 meters
- Operating temperature: -10°C to +55°C
- Liquid and dust protected (IP 65)
- Multi-functional button (alarm call, answer call, etc.)
- Text message support (requires AIWS, 30 messages sent/received storable, message length 160 characters)
- Limitations due to power constraints for Intrinsically Safe handsets:
  - Display brightness is lower
  - The keypad has no backlight for security reasons
  - The vibrator and sound are not activated simultaneously (Ringing and vibration alternate)
  - The ring signal/beeper volume is slightly lower than 3740
  - Loudspeaker volume is slightly lower lower than 3740

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	Telephone	3720 DECT Telephone	3725 DECT Telephone	3740 DECT Telephone	3749 DECT Telephone
Keys:	Keypad	yes	yes	yes	yes
	Hook on/off buttons	yes	yes	yes	yes
	Softkeys	3	3	3	3
	Navigation cluster	4-way	5-way	4-way	4-way
	Volume buttons		yes	yes	yes
	Mute button		yes	yes	yes
	Alarm button		yes	yes	yes
Display:	Display, illuminated	112 x 115 pixels, Black and white	128 x 160 pixels, 64K colors	128 x 160 pixels, Black and White	128 x 160 pixels, 64K colors
	Battery indicator	yes	yes	yes	yes
	Signal strength indicator	yes	yes	yes	yes
	Time and date display	yes	yes	yes	yes
Audio:	Handsfree	yes, 2-way speakerphone		yes, 2-way speakerphone	
	Headset interface 2.5 mm	yes		no, IP65 compatible special plug	
	Bluetooth 2.0 Headset profile	no	yes	no	yes
	Supported Ring tones	14	14	14	14
	Ring tone volume	8 levels		8 levels	
	Vibration	yes		yes	
	Ringing options	Ring, Vibration, visual		Ring, Vibration, visual	
	Usage profiles	supported		supported	
	Keyes beep	yes, selectable		yes, selectable	
	Volume control	8 levels	8 levels	8 levels	8 levels
Features	Message LED (Red-green)	yes	yes	yes	yes
	Call log (Local)	25 entries	25 entries	25 entries	25 entries
	Local Directory/Contacts	250	250	250	250
	Access to IP Office Directory	Possible[1]	Possible[1]	Possible[1]	Possible[1]
	Support of Text Messaging	no	Optional[1]	Optional[1]	Optional[1]
	Phone lock	pin protected	pin protected	pin protected	pin protected
	Keyes lock	yes	yes	yes	yes
	Programmable "in Call Feature" menu	10 functions	10 functions	10 functions	10 functions
	Programmable "Call Services" Menu	10 functions	10 functions	10 functions	10 functions
	GAP compliant	yes	yes	yes	yes
Hardware:	Battery	Lithium Polymer	Lithium-Ion	Lithium-Ion	Lithium-Ion

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	Standby/Talk time - No Bluetooth	160/16 hours	120/20 hours	150/18 hours	80/10 hours
	Standby/Talk time - with Bluetooth	n.a.	120/13 hours	n.a.	???
	Charge time	4 hours	4 hours	4 hours	4 hours
	Dust/Liquid Protection level		IP 44	IP 65	IP 65
	Ruggedness	IEC 68-2-32, procedure 1, dropped 12 times from 1 metre		IEC 68-2-32, procedure 1, dropped 12 times from 2.0 m	
	Intrinsically save design (for use e.g. in mining, chemical industry, gas-stations)	No	No	No	Yes
	Size	133 × 53 × 24	134 × 53 × 26 mm	143 x 59 x 29 mm	143 x 59 x 29 mm
	Weight	115 g (battery and clip included	130 g (battery and clip included)	180 g (battery and clip included)	180 g (battery and clip included)
	Belt clip options	none, standard, swivel		none, standard, swivel	
Localization:	Supported languages	English,	Czech	Czech	Czech
		German	Danish	Danish	Danish
		Spanish	Dutch,	Dutch,	Dutch,
		French	English	English	English
		Downloadable language (one of 14[2])	Finnish	Finnish	Finnish
			French	French	French
			German,	German,	German,
			Greek	Greek	Greek
			Hungarian	Hungarian	Hungarian
			Italian	Italian	Italian
			Norwegian	Norwegian	Norwegian
			Polish	Polish	Polish
			Portuguese (Brazilian)	Portuguese (Brazilian)	Portuguese (Brazilian)
			Portuguese	Portuguese	Portuguese
			Russian	Russian	Russian
			Slovakian,	Slovakian,	Slovakian,
			Spanish	Spanish	Spanish
			Swedish	Swedish	Swedish

1. Requires AIWS Server for integration with IP Office

2. Language supported on phone, IP Office menu items might appear in English.

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Supported Bluetooth Headsets on the 3725 and 3749 headsets:

3725 and 3749 have build in Bluetooth 2.0, supporting the "headset" profile. While others standard compliant headsets should work, Avaya has only tested a selected number.

The following headsets are recommended:

- Plantronics Voyager 520
- Plantronics Voyager 835
- Plantronics Explorer 390
- Nokia BH-804
- Nokia BH-904
- SonyEricsson HBH-PV740
- Peltor M2RX7-WS4
- SonyEricsson HBH-PV702
- SonyEricsson HBH-GV435
- SonyEricsson HBH-IV835
- Plantronics Voyager510
- Plantronics Explorer665
- Nokia BH-201
- Jabra BT8010
- Motorola H350
- Motorola H670
- Samsung WEP4102

DECT R4 handsets on IP DECT solution:

The new DECT R4 handsets will work with the legacy IP DECT system. Nevertheless not all features available with the 3701 and 3711 handsets will be available on the DECT R4 handsets as well. An (incomplete) list of working features is:

- Subscription with PARK.
- Basic Call.
- Display Messages (with some limitations, e.g. truncation at end of line).
- Handover.
- Roaming.
- R-Key Handling (Enquiry call, Conference, ...).
- Feature Access Codes.
- Distinguish between internal/external calls.
- DTMF during call.
- Local call log (calling party numbers are transmitted).
- Local time and date.

Features known to be networking with DECT R4 handsets and the legacy IP DECT system are

- Access of IP Office/ external directory (e.g. LDAP).
- Get time and date from the system.
- WML (WAP access).
- SMS/Messaging (not a feature of the legacy IP DECT system).

Generally it is highly recommended to do a sample installation with one handset first before rolling out a larger set of DECT R4 handsets on a legacy IP DECT system

3701 IP DECT

This telephone is fully supported on the Avaya IP DECT system. It is no longer available for new sales.

However, it can also be used on the Avaya DECT R4 system in 'GAP-mode' with functionality limited to standard calling functionality, DECT roaming and handover. The built in feature list will not work on DECT R4.

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- Listen-only handsfree speaker.
- SOS Emergency key for speed dialing an emergency number.
- Information key that can be used for:
  - Phone number lists and voice mail indication.
  - Information and speaker key flash when active.
- 50 phone book entries in every handset
- 10 possible ring tones with temporary mute.
- 4-level signal strength display.
- Speaker and handset volume, 3-levels and mute capability.
- Manual and automatic key lock (1 minute timer).
- Temporary ring tone muting.
- Silent charging.
- 12 menu languages: Czech, Danish, Dutch, English, Finnish, French, German, Italian, Norwegian, Portuguese, Spanish and Swedish. However, in the Czech and Norwegian language mode some menu items may appear in the English language.
- Illuminated 3-line graphic display (96 x 33 pixels), variable 3-level contrast.
- Stand-by time: Up to 100 hours.
- Talk time: Up to 10 hours.
- Charge time: max. 6 hours for empty batteries.
- Weight: 138 grammes including 3 AAA (NiMH) batteries.
- Dimensions (Height x Width X Depth): 146 x 55 x 28 mm.

Optional telephone accessories include:

- Desktop charger.
- An adapter cord for use with headsets.
- Heavy-duty belt clip.

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3711 IP DECT

This telephone is fully supported on the Avaya IP DECT system. It is no longer available for new sales.

However, it can also be used on the Avaya DECT R4 system in 'GAP-mode' with functionality limited to standard calling functionality, DECT roaming and handover. The built in feature list will not work on DECT R4.

The 3711 phone supports the same features as the 3701 IP DECT handset but with the following differences:

- Full handsfree speakerphone operation.
- Headset connection (2.5 mm jack).
- Vibrating alarm.
- Personal phone book with 100 entries
- Access to system phone book.
- Voice Mail indication.
- Choice from 30 ring tones.
- Speaker and handset volume, 7-levels and mute capability.
- Automatic call pick-up using a headset.
- 10 menu languages: Danish, Dutch, English, Finnish, French, German, Italian, Portuguese, Spanish and Swedish.
- Illuminated 5-line graphic display, (96 x 60 pixels), variable 7-level contrast.

Optional handset accessories include:

- Desktop charger.
- An adapter cord for use with headsets.
- Heavy-duty belt clip.

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Digital Wireless 3810 Telephone

Features

- 2-line, 32 character Handset Liquid Crystal Display (LCD).
- 10 hours of talk time, and 4 days of standby time.
- 4 displayed operation modes indicating Talk, Ringer On/Off, Battery Low, and Message Waiting.
- Single button access to fixed features – Hold, Transfer, Conference, and Redial.
- 4 programmable buttons to access features on the PBX.
- 20 Number Memory for quick and easy speed dialing
- 10 channels, supporting up to 10 simultaneous conversations in overlapping radio coverage areas.
- Headset jack.
- Ringer and Handset volume control.
- User selectable ring type.
- Vibrate alert.
- Redial Button
- Base Unit and Charger Unit.

The Avaya 3810 Wireless Telephone is a digital telephone designed to work with IP Office from release 2.0 and above by connecting to a Digital Station (DS) port. It offers the mobility inherent in a wireless telephone plus access to a number of features and functionality of the connected communications system.

A maximum of 5 Avaya 3810 wireless handsets can be connected to the same IP Office in any overlapping radio coverage area.

This phone is only available in the USA and Canada.

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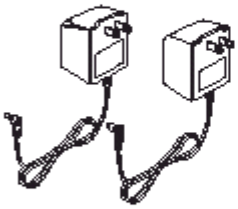
The Avaya 3810 is delivered as a single unit containing:

- Base Unit.
- Handset.
- Telephone Cord.
- Base Unit Power Supply Adapter.
- Charging Stand Power Supply Adapter.
- Rechargeable Battery.
- Belt Clip.
- Charging Stand.
- User and Installation Guide.
- Wall Plate Adapter.

Base Unit

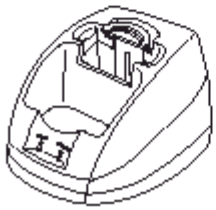
Handset

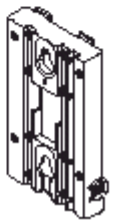
Telephone Cord


2 AC Adapters

Rechargeable Battery

Beltclip


Charger Unit


Wall Mount Stand

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Business Series Terminal T7406E Cordless Telephone (not supported in EMEA)

The T7406E Cordless Telephone is a desk-centric mobility solution, designed for use in a small- to medium-sized enterprise site, branch office or franchise site. The T7406E Cordless Telephone enables businesses to deliver better customer service while making employees more productive with workspace mobility. The T7406E is supported on IP Office modules also supporting the BST digital sets.. The T7406E is available in North America, Mexico and Caribbean countries excluding Jamaica and Trinidad.
.

Features

An integrated multiline phone, the T7406E leverages the features of the popular Business Series Terminals portfolio and has the following capabilities:

- 2.4 GHz ISM Band
- WiFi-“friendly” technology that enables the product to work in the vicinity of a wireless LAN with no interference
- Frequency-hopping spread spectrum (FHSS) available on 90 channels for each base
- Operating range of 350 meters (1000 feet) in outdoor clear line-of-sight
- 64-bit standards-based security encryption
- Cellphone-like features such as a personal directory, key lock, handsfree speaker, back-light display and keypad, vibration notification
- Maximum site density is two (2) base stations, each accommodating up to four (4) handsets
- Nickel metal hydride battery delivers 4 to 5 hours of talk time or 36 hours of standby time
- Convenient features such as speaker phone, vibrate mode, 20-name directory, out-of-range alert, message waiting and signal strength indicator
- Accessories include a belt clip, handset charging cradle, external battery charger, and AC adaptor

Digital Mobility Solution

The Digital Mobility Solution is a modular wireless voice communication system that is based on the Digitally Enhanced Cordless Telephony (DECT) technology. Supported on IP Office from Release 7, the Digital Mobility Solution scales from 1 to 64 users and covers an area up to 1.5-million square feet for true campus-wide mobility. The solution consists of the following components:

- Digital Mobility Controller 080/081 (DMC 080 supported in North America; DMC 081 supported elsewhere)
- Digital Mobility Controller 320/321 (DMC 320 supported in North American; DMC 321 supported elsewhere)
- Basestation (4 channel)
- Repeater (2 channel or 4 channel)
- Handsets (3X16 displays)
- Handset Accessories

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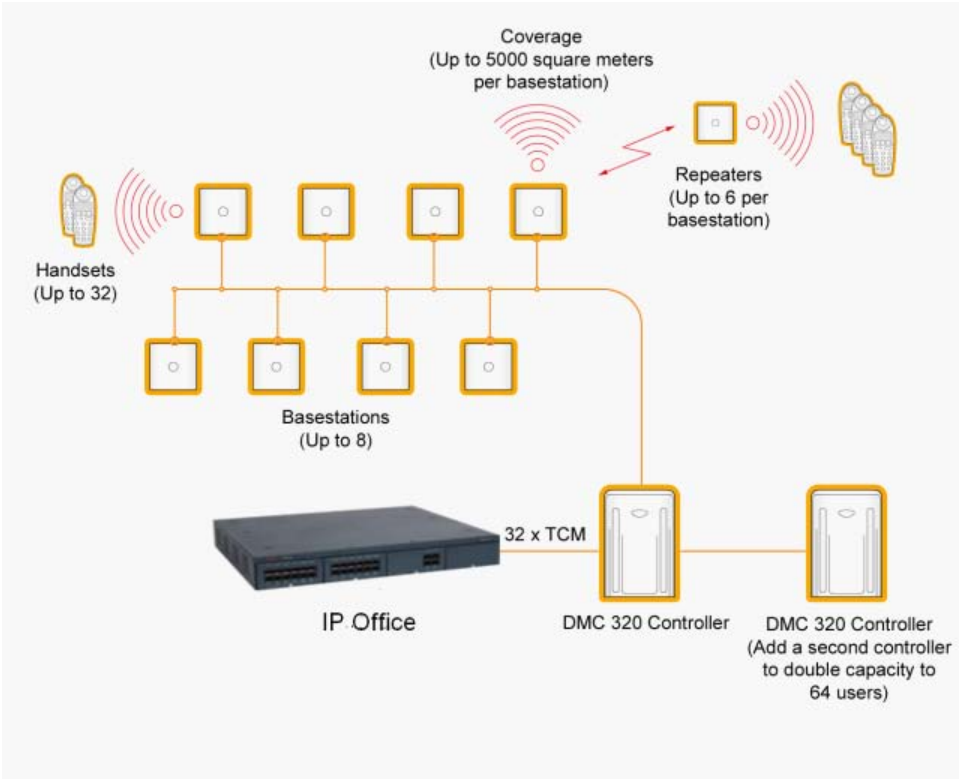
- OA&M Package
- Service Tool (H/W & S/W) for handsets and repeaters
- Deployment Tool

Figure 1.



Connection to IP Office

Connection to IP Office is achieved by using one digital line per DECT phone. The same digital line interfaces are required that also support the BST digital telephones like the T7000 series of telephones.

Digital Mobility Controllers

The following controllers are available within the Digital Mobility offering:

- Digital Mobility Controller 080/081 (DMC 080/081)
- Digital Mobility Controller 320/321 (DMC 320/321)

The Digital Mobility offering has a multi-cell option that allows you to link two DMC units together to effectively double the capacity, delivering up to 5,000 square meters coverage per base station. DMC 080 and 320 are supported in North America. DMC 081 & 321 are supported elsewhere.

Figure 2.Digital Mobility Controller 080/081 (DMC 080/081)

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Figure 3.Digital Mobility Controller 320/321 (DMC 320/321)

The two levels of controllers allow customers to start small and grow their mobility solution to meet their needs today and tomorrow without being charged for additional equipment until the need arises.
The DMC 080/081 will support up to eight handsets and two base stations. Two DMC 080/081 modules can be linked together, if incremental growth is required (2 DMC 080/081 linked together will support 4 base stations and 16 handsets). The DMC 320/321 will support 32 handsets and 8 base stations. The DMC 320/321 can also be linked to expand up to 64 handsets and 16 base stations.
The DMC module provides power to each base station via twisted pair cable. The base station can be located up to 4921 feet (1500 m) from the DMC module. The two-channel repeater is used to extend coverage from a base station, up to 50% additional coverage. Up to six repeaters can be registered to a specific base station to extend the range of a base station. The repeaters are linked to the base station in a wireless manner, so that no physical connection is required. Local power is required for each repeater.
The solution does offer an external antenna which can be used in conjunction with a repeater to extend a base station signal up to 0.62 miles (1 km) away. A typical application would be a campus with multiple buildings requiring coverage.

Table 1.Coverage Area

Digital Mobility Solution Configuration	Maximum Number of Base stations	Maximum Number of Users	Maximum Number of Simultaneous Calls	Maximum Coverage: Square Feet	Maximum Coverage: Square Meters
DMC080/081	2	8	8	180,000	20,000
DMC080/081 + DMC080/081	4	16	16	360,000	40,000
DMC320/321	8	32	32	720,000	80,000
DMC320/321 + DMC080/081	10	40	40	900,000	100,000
DMC320/321 + DMC320/321	16	64	64	1,440,000	160,000

Coverage depends on the environment (including building material and equipment)

Digital Mobility Base Stations

Each Digital Mobility base station supports up to four simultaneous calls and provides up to 5,000 square meters of coverage (depending on environment). One base station can support up to six repeaters to further extend the Digital Mobility coverage.

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Figure 4.Digital Mobility Base Station

Digital Mobility Repeaters

Each Digital Mobility repeater provides two or four channels and supports two or four simultaneous conversations while extending coverage from a base station by providing up to 50 percent additional mobility coverage. Up to six repeaters can be registered to a specific base station in order to further extend base station range. Repeaters are linked to the base station wirelessly, so no physical connection is required. Local power is required for each repeater.

The Digital Mobility solution also includes an external antenna which can be used in conjunction with a repeater to extend a base station signal up to one kilometer away. A typical application would be a campus with multiple buildings requiring coverage.

Figure 5.Digital Mobility Repeater

Digital Mobility Handsets

A portfolio of handsets are supported with the mobility portfolio:

Digital Mobility Handset 7430

A high value and low cost handset fulfilling all basic requirements for a mobility handset. Targeted at industry, warehouse, and retail segments. The 7430 is available in North America, and Argentina only. Equivalent handsets include the 4135 (EMEA and Hong Kong) 4136 (Australia and New Zealand), and 7434 (South America).

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Digital Mobility Handset 7449

A classic high-quality rugged handset with a high level of functionality, targeted at office, hospital, and customer support centers. This handset has an IP 54 classification (protection against dust and splashing water).
The 7440 is available in North America, and Argentina only. Equivalent handsets include the 4145 (EMEA and Hong Kong) 4146 (Australia and New Zealand), and 7444 (South America).
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Digital Mobility Handset 4145Ex

The ultimate handset with IP 54 and intrinsic safe classifications. Targeted for use in explosives environments within oil, gas and chemical production.
The 4145Ex is available in EMEA. Equivalent handsets include the 4146Ex (Australia and New Zealand).

Common Handset Attributes

- IP Office feature integration similar like the BST telephones, including 3 soft keys
- Graphical display
- Auto Login - Roaming between 4 different systems
- Headset jack (not included with 7430/4135)
- Loud speaker (Hands free mode) –7440/4145 only
- Telephone book with storage for 80 names and numbers
- Can be set to vibrate mode (not included with 7430)
- Text message capability
- Silent mode (mute all sounds)
- Redial function (last 10 numbers)
- Adjustable volume
- Nine different ring tones
- Key lock
- Microphone mute
- Caller-ID presentation
- Automatic hook-off
- Standby/speech-time: 100/10 hours
- Temperature compensated charging
- Weight including battery is 4.3 ounces (121 grams)
- Size: 5.6” x 1.9” x 1.0” (143 x 48 x 26 mm)

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Digital Mobility Software

The Digital Mobility Solution is supported by an easy to use Operations, Administration, and Management (OA&M) software package, a Mobility Software Service Tool, and a comprehensive Deployment Tool.

Operations, Administration, and Management Software Package

The OA&M software package allows the following aspects of the Digital Mobility solution to be managed:

- Digital Mobility Controller administration
- Handset registration and subscription
- System backup and restore programming information
- Statistical package for debug
- Broadcasting test messaging
- Updating base station firmware
- Remote system administration via RS232. Optional serial-IP converter

Mobility Software Service Tool

The Mobility Software Service Tool allows the following aspects of the Digital Mobility solution to be managed:

- Repeater programming
- Handset firmware upgrading

Note: Use of this tool requires a programming cable and service tool handset cradle.

Deployment Tool

The Deployment Tool can be used to assist with challenging deployments. For normal deployments a handset can be used to deploy the Digital Mobility Solution. Each handset can be used to perform the following administrative activities:

- General deployment requirements
  - Register master handset
  - Register and subscribe handsets
  - Subscribe multiple systems
- Base station/Repeater registration

The Deployment Tool and the ability of the wireless handsets to provide coverage information provide effective tools for quick, sure installations and coverage with minimal gaps or costly overlapping. The Deployment Tool provides superior deployment to competitive offerings and will ensure a more cost effective, more accurate installation.

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VPN Phone Functionality

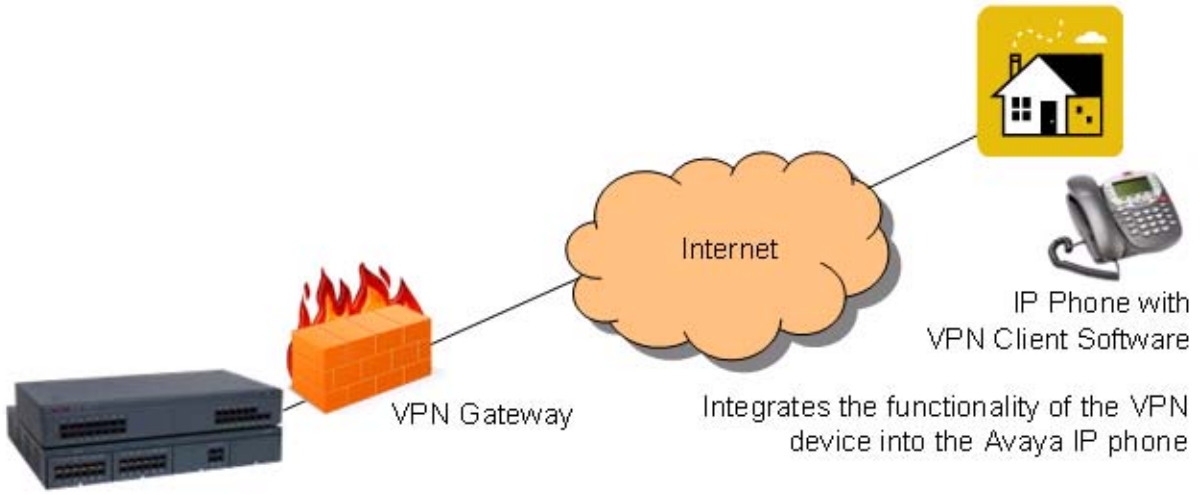
VPN Phone is a full-featured IP Telephony solution that provides secure communication over public ISP networks to an IP Office system at the company headquarters.

It is a software-only product that runs on the standard 5610/5620/5621 or 4610/21 IP telephones. In combination with one of these telephones and the most popular VPN gateway products, the software extends enterprise telephony to remote locations.

VPN functionality is also supported on the new 9600 IP telephones, not requiring a separate software load.

VPN telephones offer the full IP Office telephony features that are available on IP Office IP telephones at the users desktop in a remote location like a home-office:

From release 6 onwards, no license for VPN functionality is needed for either 5600 or 9600 Series telephones.



VPN Phone is ideal for IP Office customers supporting "work-at-home" users:

- Virtual Office workers
- Remote workers
- Remote call center
- Business continuity support
- Very small locations that require a single phone only
- Temporary installations such as conferences, off-site meetings, and trade shows

VPN Phone has been tested with a number of VPN-gateways from major vendors like Cisco or Juniper as well as with smaller VPN-access devices from companies like Adtran, Kentrox, Netgear, and SonicWall. Refer to the support pages (support.avaya.com) for a list of available application notes on VPN-gateways tested with each line of telephones.

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Analog Telephones

As well as providing a lower cost alternative to system specific telephones, analog telephones can still deliver a high degree of functionality on IP Office. They are particularly appropriate in applications where users require lower entry costs and can be used with Phone Manager for a high proportion of call control.

Analog telephones that are compatible with caller display functionality can display the telephone number of the calling party if available. Simple programming of IP Office can convert that numeric display in to the company name associated with that number.

Feature activation by analog telephones is via short codes. IP Office is pre-programmed with a default set of short codes but these can be changed to mimic a legacy telephone system as required.

Avaya would like to stress that although most analog telephones will work on IP Office - Avaya cannot guarantee that all analog telephones in every region of the world will work on the IP Office.

- Analog telephones connect to IP Office via ports marked PHONE ports.

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Avaya 6200 Analog Telephone (North America)

The 6200 range of telephones are single-line analog telephones that require one tip and ring pair for operation. This series of telephones have a Ringer volume control on the side of the telephone and a Handset volume control on the front of the phone. They use DTMF dialing only and support the Positive Disconnect function. In addition, these telephones have a Message light, a recall button that allows access to system features, a redial button that allows automatic redial, a hold button with a single associated light, and a data jack on the rear of the telephone. The 6219 phone adds 10 programmable dialing buttons and the 6221 phone adds a built-in speakerphone with mute capability.

Analog Telephone Features	6211	6219	6221
Programmable buttons (10 buttons)	✗	✓	✓
Program Keylock	✗	✓	✓
Pause	✗	✓	✓
Redial	✓	✓	✓
Speaker	✗	✗	✓
Flash	✓	✓	✓
Hold (with indicator light)	✓	✓	✓
System Hold	✗	✓	✓
Mute	✗	✗	✓
Handset Volume Control	✓	✓	✓
Ringing Volume Control (3 position)	✓	✓	✓
Ringing Patterns (2)	✓	✗	✗
Personalized Ringing	✗	✓	✓
Message Waiting Light	✓	✓	✓
Desk/Wall Mount	✓	✓	✓
Data Jack	✓	✓	✓
Colors	White/Grey	White/Grey	White/Grey
Ringer Equivalency	0.7A, 1.6B	0.5A, 1.5B	0.5A, 1.5B
Hearing Aid Compatible	✓	✓	✓
Positive Disconnect	✓	✓	✓
DTMF Dialing	✓	✓	✓
Specialty Handset Support	✓	✓	✓

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Interquartz Gemini Phones (EMEA and APAC)

Avaya have tested the new generation Interquartz Gemini analog telephones with IP Office to ensure that telephone and system are compatible. The Gemini telephones offer good value for money without compromising on quality. Their stylish design and rugged build quality make them a popular choice for buyers on a limited budget.

For sales enquiries and product information contact Interquartz at avaya-enquiries@interquartz.co.uk.

Basic telephone 9330-AV

- Visual Message Waiting Indication.
- Locking mute button with LED indicator.
- Last number redial.
- Recall button.
- Ringer volume adjust.
- Ringer indicator light.
- Wall mountable - no additional bracket required.
- Hearing aid compatible.
- Rubber feet to minimize slippage

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All features of 9330-AV plus:

- Caller ID with 80 memories (shows date, time and new/repeat/answered/unanswered calls) .
- Large 3 line LCD display.
- IP Office feature activation through programmable keys.
- 100 name and number personal directory.
- 20 lockable direct access memories.
- Full handsfree working.
- Headset port.
- Switchable Time Break Recall 100 / 200 / 300 / 600 ms.
- Call timer.
- Alphanumeric keypad.
- Last number redial with 5 memories.

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- Removable inlay card for personalized logo printing.
- Triple standard message waiting light (high voltage, reverse polarity and voltage drop).
- 10 non-volatile memories.
- Ringer indicator light.
- Ringer volume and pitch adjustment.
- Last number redial and Recall button.
- Hearing aid compatible.
- Wall mountable – no additional bracket required.
- ELR/TBR switchable.
- MF Only.

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Doorphone Entry Systems for IP Office

Doorphones offer convenience and security. Depending on the needs of the environment, door telephones may allow internal users to not only speak with someone who is outside, but also to easily allow the visitor entrance to the facility or residence. Doorphones can be connected to the Avaya IP Office base unit in a variety of ways, providing design flexibility based upon needs.

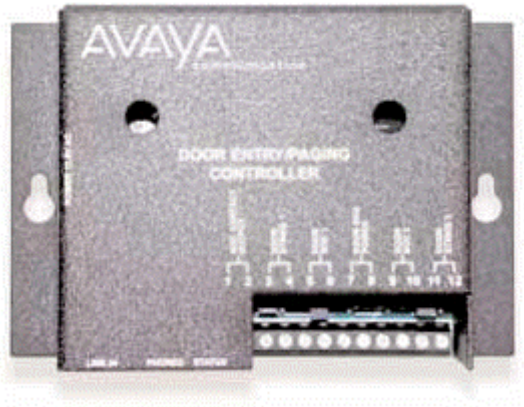
All of the IP Office base units include an external output port. Connections of doorphones to these ports enable the user to gain

access to the premises through default system short codes, through the optional Phone Manager Pro application, and through the optional VoiceMail Pro application. The flexibility of the IP Office provides the ability for short codes to be customized to a code more desirable for users. By using the Phone Manager Pro application, users can label the icons within the application a descriptive name such as Receiving Door or Front Door. The flexibility of VoiceMail Pro allows the visitor to enter a predetermined code from the phone granting access. This scenario is particularly useful in areas when co-workers are working at another site. Additionally, many doorphones can be connected to station or trunk ports available on IP Office.

The Avaya IP Office system offers three doorphone solutions to choose from:

- Avaya Universal Doorphone System (North America)
- Kalika Communications Doorphone Entry System (EMEA)
- Interquartz Doorphone (EMEA)

Avaya Universal Doorphone System:



- System consists of a controller and a speaker.
- The speaker is mounted securely on the wall and is connected to the controller, which normally resides in the equipment room. The controller is connected to a trunk port.
- Users with the trunk appearance will be notified when a visitor has pressed the Push button located on the weatherproof speaker.
- Each controller supports two speakers, for example Front Door and Back Door.
- Custom ringing mode distinguishes doorphone calls from external calls.
- Call waiting tones indicate which doorphone is calling and distinguish a doorphone call from an external line call.
- Calls can be placed on hold when visitors call from the doorphone.
- Commercial or residential security is provided via two-way handsfree communication from a door or gate.

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Headsets

Avaya offers ergonomically designed communication headsets and amplifiers for the Avaya IP Office telephones. This full line of professional and contact center solutions set the standard in sound quality and durability. Avaya headsets are designed for maximum, all-day comfort and are available in styles that suit nearly any wearer and any usage pattern.

Whether you want the freedom to communicate handsfree while working at your desk, or the ability to roam while talking, you will find a solution that suits your individual needs.

Please check www.avaya.com for a current list of supported headsets.

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Summary

All Avaya telephones are designed to ensure that features and functions are easily accessible to the user - ensuring that, through ease of use, the full benefits of the system are delivered to the desktop.

The telephones that support handsfree operation are intended for individual use only, not for group and conference room operation.

The telephones listed below are the preferred and premier range of telephones for use on the IP Office. These telephones are sold worldwide in every country that the IP Office is available. This telephone range consists of both digital and IP telephones.

IP Office worldwide digital telephones:	IP Office worldwide IP telephones:
• 1403 Telephone	• 1603 IP Telephone
• 1408 Telephone	• 1608 IP Telephone
• 1416 Telephone	• 1616 IP Telephone
• 9504 Telephone	• 9608 IP Telephone
• 9508 Telephone	• 9621 IP Telephone
	• 9641 IP Telephone

In addition to the telephones above, the IP Office supports a wide range of telephones as listed below. However, note that some of these telephones are only available in certain countries and regions.

Other digital/analog telephones:	Other IP telephones:
• 5402 Telephone	• 9620L/9620C IP Telephone
• 5420 Telephone	9630G/9640/9640G IP Telephone
• 5410 Telephone	9650/9650C IP Teleph
• T3 Comfort (Up)	• 3720 DECT R4 Wireless Handset
• T3 Classic (UpnP)	• 3725 DECT R4 Wireless Handset
• T3 Compact (UpnP)	• 3740 DECT R4 Wireless Handset
• T3 DSS Module	• 3749 DECT R4 Wireless Handset
• 2402 Telephone	• 3810 Wireless Telephone
• 2410 Telephone	• 36451 Wireless (WiFi) Phone
• 2420 Telephone	• 3645 Ruggedized Wireless (WiFi) Phone
• EU24/EU24BL DSS Unit	•

The following Norstar/BCM telephones are fully supported on IP Office R7 but might require special interface cards/modules. Not all of them are available for new sales and they are not available to all channels.

Nortel Heritage Digital telephones	Nortel Heritage IP telephones:
· T7000	· 1120 IP phone
· T7100	· 1140 IP phone
· T7208	· LCD KEM for 1100 telephones
· T7316; T7316e	· 1220 IP phone
· T24 KIM– Key Indicator Module	· 1230 IP phone
· M7000	· LCD KEM for 1200 telephones
· M7100	· LED KEM for 1200 telephones
· M7208	· 7420/7430/7440 Digital Mobility Handsets (2.4 GHZ)
· M7310	· 7439/7449 Digital Mobility Handset, DECT, North America
· M7324	· 4135/4145/4145EX Digital Mobility Handset, DECT; EMEA and HongKong
· CAP (48 buttons) – Central Answering Position	· 4136/4146/4146EX Digital Mobility Handset, DECT;Australia and NZ.
· T7406, T7406E BXT cordless Telephone, North America	· 7420/7430/7440 Digital Mobility Handsets, DECT, South America

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The following telephones are fully supported on IP Office R6 but are no longer available for new sales.

Other digital/analog telephones:	Other IP telephones:
	• 5601 IP Telephone
	• 4601 IP Telephone
	• 4602SW IP Telephone
	• 5602SW IP Telephone
	• 4610SW IP Telephone
• 4406	• 5620 IP Telephone
• 4412	• 4620/21 IP Telephone
• 4424	• 4625SW IP Telephone
• 4450 DSS Unit	• 3616 Executive Wireless (WiFi) Phone
• 5610SW IP Telephone	•5621 IP Telephone
4610SW IP Telephone	• 3620 Healthcare Wireless (WiFi) Phone

The following Norstar/BCM Digital mobility components are supported on IP Office R7 but might require special interface cards/modules. Not all of them are available for new sales and they are not available to all channels

Digital Mobility 2.4 GHz solution - North America and CALA	
Digital Mobility Base Station 10	(2.4GHz, Compatible with Handset 7430/7440, Repeater 40)
Digital Mobility Repeater 40	(2.4GHz, Compatible with Handset 7430/7440, Base station 10)
Digital Mobility External Antenna 51	
DMC 080	Digital Mobility Controller - N.A.
DMC 320	Digital Mobility Controller - N.A.
Digital Mobility 1.9 GHz solution - North America (1929-1930 MHz)	
Digital Mobility Base Station 19 (NAR)	
Digital Mobility Repeater 49 (Canada, US)	
Digital Mobility External Antenna 51	
DMC 080	Digital Mobility Controller - N.A.
DMC 320	Digital Mobility Controller - N.A.
Digital Mobility 1.8 GHz solution - Europe, Hong Kong, Australia, New Zealand (1880-1900MHz)	
Digital Mobility Base Station 15	
Digital Mobility Repeater 25	
Digital Mobility Repeater 45	
Digital Mobility External Antenna 51	

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DMC 081	Digital Mobility Controller - International
DMC 321	Digital Mobility Controller - International
Digital Mobility 1.8 GHz solution - South America (1900-1930 MHz)	
Digital Mobility Base Station 14	
Digital Mobility Repeater 24	
Digital Mobility External Antenna 51	
DMC 081	Digital Mobility Controller - International
DMC 321	Digital Mobility Controller - International

The following telephones are fully supported on IP Office R6 but are no longer available for new sales.

Other digital/analog telephones:	Other IP telephones:
	• 5601 IP Telephone
	• 4601 IP Telephone
	• 4602SW IP Telephone
	• 5602SW IP Telephone
	• 4610SW IP Telephone
• 4406	• 5620 IP Telephone
• 4412	• 4620/21 IP Telephone
• 4424	• 4625SW IP Telephone
• 4450 DSS Unit	• 3616 Executive Wireless (WiFi) Phone
• 5610SW IP Telephone	•5621 IP Telephone
4610SW IP Telephone	• 3620 Healthcare Wireless (WiFi) Phone

- For maximum cabling distances please refer to the IP Office Installation Manual.

Legacy Telephone Support

The following telephones are not supported by IP Office Release 6. They may function but have not been tested with IP Office Release 7.0 and any faults reported with IP Office Release 7.0 will not be fixed.

- 20DT Analog DECT used with IP Office Analog DECT and Compact DECT.
- TransTalk 9040

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Chapter 6. Features

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6. Features

IP Office provides a comprehensive telephony feature set to enable a fast and efficient response to a telephone call. Features such as Caller ID display and call tagging allow employees to see who is calling and who they are calling before they pick the call up. Client information can even be 'popped-up' on the user's PC.

For those who are not tied to a desk, Wireless handsets and twinning offer mobility around the office. For those out of the office, be it on the road or working from home, comprehensive and easy to use call forwarding facilities, PC Softphone and a remote access service allow them to remain in telephone contact and access centralized resources at all times.

Incoming calls can be efficiently handled using either Direct Dialling (DDI/DID) or dedicated operators. For out of hours calls or times when you just can't take calls, IP Office provides voicemail and optional Auto-Attendant services.

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Basic Call Handling

Tones

IP Office generates the correct user tones for the geography. These tones are generated for all IP Office extension types, analog, digital and IP.

Supported tones are:

- Normal, alternate, and secondary; depending on geography
- Busy
- Unobtainable
- Re-order
- Conferencing tone depending on geography

Caller ID

Feature

- Display of the caller's number on incoming calls, where supplied by the service provider.
- Sending of calling number on outgoing external calls.

Benefit

- Confirmation and recognition of who is calling.
- Storage of Caller ID numbers for return calls.
- Directory name matching to Caller ID numbers.
- Screen-Popping customer records in compatible applications.

Description

Where supplied by the service provider, the IP Office can receive and use the caller's Caller ID. The Caller ID is passed through to the answering phone or application and is included in any call log or history supported by the phone or application. If the Caller ID matches a number in the IP Office's Directory, the matching directory name is shown instead of the number.

Where IP Office Phone Manager, or the TAPI service is used to link to database software on the users PC, it is possible to have an automatic query performed on the supplied Caller ID and have the caller's record in front of the user before the call is answered.

For outgoing calls the IP Office can insert a system wide Caller ID or set a flag to have Caller ID withheld. For users with a direct dial number routed to their extension, that direct dial number is also used as their Caller ID for outgoing calls. Alternatively short codes can be used to specify the Caller ID that should be sent with outgoing calls.

Note that the sending and receiving of Caller ID is subject to the service provider supporting that service. The service provider may also restrict which numbers can be used for outgoing Caller ID.

Hold

A call may be placed on hold with optional Hold music. A held call cannot be forgotten as it is presented back to the extension after a timeout set by the system's administrator.

See also Park 202.

Toggle Calls

Toggle Calls cycles round each call that the user has On Hold to their extension locally within the system, presenting them one at a time to the user.

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Hold Call Waiting

Hold Call Waiting is a compound feature combining hold and answer and provides a convenient way to hold an existing call and answer a waiting call through a single button press.

Hold Music (Music on Hold)

The IP Office system supports up to 4 sources of music on hold – system sources which may be external: one system source plus 3 additional external sources using analog extensions. Internal (WAV) or tone plus up to 3 additional internal sources. The internal sources are .WAV files saved either in volatile memory, or on the optional memory card in an IP500 and IP406 V2. The .WAV file must be 16bit PCM mono and sampled at 8Khz with a maximum duration of 30 seconds.

Alternate sources for music on hold are selectable for use by Incoming Call Routes or Hunt Groups. On IP500, and IP500 V2 systems, each source can be up to 90 seconds long.

External music on hold sources connect to the 3.5mm Audio socket on all IP Office control units, or configured analog extensions.

Park

As an alternative to placing a call on hold, a call can be parked on the system to be picked by another user.

The call park facility is available through the user's telephone, Avaya one-X™ Portal for IP Office, Phone Manager or SoftConsole. Calls are Parked against a 'park slot number' which can be announced over a paging system so the person the call is for can go to any phone and collect the call by dialling the park slot number.

For convenience Phone Manager has 4 pre-defined park buttons. On digital telephones with DSS/BLF keys it is possible to program Park keys that will indicate when there is a call in a particular park slot and allow calls to be parked or retrieved.

There is a system configurable timeout that determines how long a call may remain parked before it is re-presented to the extension that originally parked the call.

Automatic Callback

Feature

- When calling an extension that is busy, set the system to call you when the extension becomes free. This feature is also called "Ringback When Free".
- When calling an extension that just rings, set the system to call you when the extension is next used. This feature is also called "Ringback When Next Used".

Benefit

- Carry on with other work and let the system initiate a call for you when the extension becomes available.

Description

Depending on the type of phone a user has, call back when free is accessed by dialing a short code if an internal busy tone is heard, selecting an option from an interactive menu or pressing a programmed DSS/BLF key. Callback when free can also be activated from Phone Manager.

You can also set a callback when free or a callback when next used using a short code without attempting a call.

This feature is supported across the IP Office Multi-Site/Small Community Network 257.

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Direct Inward Dialing (DID/DDI)

This relies on the local telephone exchange passing all or part of the dialed number to the IP Office. This number can then be used by IP Office call routing software to route the call to an individual phone, or groups of telephones. This service is typically used to reduce the workload on a reception position by giving members of staff or departments individual numbers so they can be called directly. For convenience it is common to have the extension or group number the same as the digits supplied from the network, but IP Office can convert the number to a number as needed by the business, within limits

In North America, T1 circuits are required for DID.

Transfer

Call Transfer allows users to transfer a call in progress to another phone number – either internal extension or external public number. The caller is placed on hold while the transfer is performed.

If the phone is put down before the destination has answered, the original caller will be automatically transferred. This is called an Unsupervised or Blind Transfer. Alternatively, a user can wait for the destination to be answered and announce the transfer before hanging up to complete the transfer. This is called a Supervised Transfer.

Unless restricted by the system administrator, the IP Office makes no differentiation between internal or external call transfers.

Distinctive and Personalized Ringing

The IP Office uses different ringing sequences to indicate the type of call, for example whether internal or external. This feature is called 'distinctive ringing'. For analog telephones the distinctive ringing sequences used are adjustable. For digital and IP telephones the distinctive ringing sequences are fixed as follows:

- Internal Call: Repeated single-ring.
- External Call: Repeated double-ring.
- Ringback Call: Single ring followed by two short rings.

This ring is used for calls returning from park, hold or transfer. It is also used for call back when free and voicemail ringback calls.

This feature is supported across the IP Office Multi-Site/Small Community Network 257.

Personalized Ringing

In IP Office the term personalized ringing is used to refer to changing the sound or tone of a phone's ring. On many Avaya digital telephones, the ringer sound can be personalized. Changing the ringer sound does not alter the ring sequence used for distinctive ringing. This feature is local to the telephone and not supported on all types of telephones.

Message Waiting Indication

Message waiting indication (MWI) is a method IP Office uses to set a lamp or other indication on compatible telephones when a new message has been left for the user, either in a personal voice mailbox or in a group mailbox or call back message. When the message has been played or acknowledged, the lamp is turned off.

All Avaya digital and IP telephones have in-built message waiting lamps. one-X Portal for IP Office provides message waiting indication on screen.

For analog telephones, from IP Office 3.1 a variety of analog message waiting indication (MWI) methods are provided. Those methods are 51V Stepped, 81V, 101V and Line Reversal. The MWI method must be selected from the IP Office Manager when configuring a system to match the properties of the analog telephones. Note that the 101V signaling is only available on IP500 Phone cards and expansion modules plus version 2 IP400 Phone 8, 16 and 30 expansion modules.

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Visual Voice

Feature

- Provides interface to voicemail through display and buttons e.g. Listen, Save, Delete, Fast Forward.

Benefit

- Quick access to voicemails and commonly used messaging features.

Description

You can access and control voice messages via the display on Digital or IP telephones. Visual Voice requires Preferred Edition or Essential Edition, and can only be used with large display LCD sets only from the 1400, 1600, 2400, 5400, 4600, 5600, 9500, 9600 and T3 series. (1403, 1603, 1603SW, 2402, 5402, 4601, 4602SW, 5601, 5602SW do not support Visual Voice). On telephones that have a display but do not support visual voice operation, use of the button for user mailbox access using voice prompts and for direct to voicemail transfer during a call is supported (does not include T3 and T3 IP telephones.

Features supported are:

- access new/old/saved messages for personal and hunt group mailboxes.
- next and previous message.
- fast forward and rewind.
- pause message.
- save, delete and copy message to other users of the system.
- change default greeting.
- change password.
- change email settings (Preferred Edition only).

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Advanced Call Handling

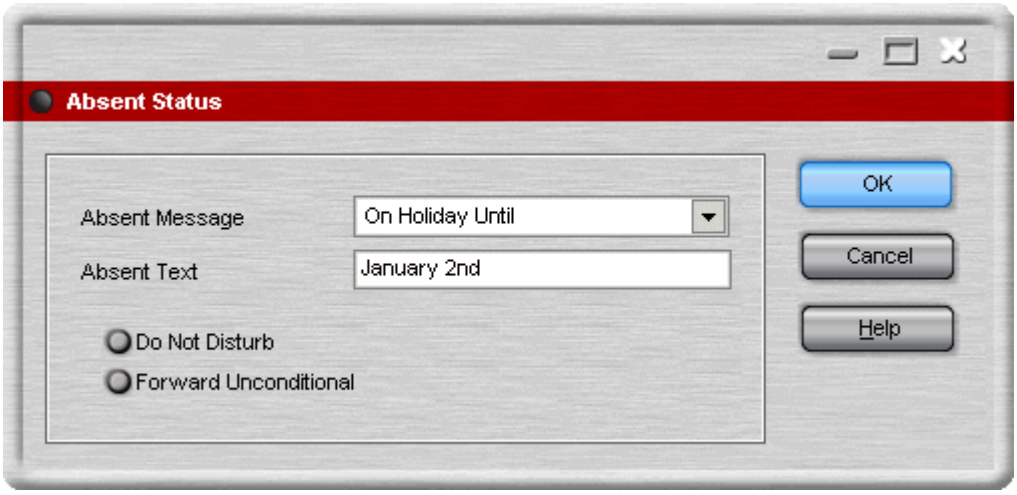
Description
In larger businesses or businesses with greater reliance on the telephone for internal and external communications some of the more advanced features will improve efficiency and customer service. Features like Pick-Up which permit users to take a call for a colleague who is temporarily away from their desk, or Absence Text which can quickly give information to internal callers about a person's availability.

Absence Text

Feature
- Display a text message on the user's phone and IP Office Phone Manager application.
- Display the same message on other internal telephones and IP Office applications when calling the user.

Benefit
- Inform other internal users of your current status and likely availability.

Description
Any user can set Absence Text on their phone, even users of standard analog telephones, but it can only be displayed on selected display telephones, Phone Manager and SoftConsole that call the user. Most supported feature telephones give the option of adding some text, for example, "At lunch until 16:00".



When a user has an absence text message set, call processing is not affected to the user and they still have the choice of using features like Do Not Disturb or Forward on No Answer as appropriate. telephones that support the interactive setting of Absence Text will also display it on the users own phone for the benefit of people who come to their desk. There are 10 predefined strings for Absence Text:

- None (no text message)
- "On vacation until"
- "Will be back"
- "At lunch until"
- "Meeting until"
- "Please call"
- "Don't disturb until"
- "With visitors until"
- "With cust. til"
- "Back soon"
- "Back tomorrow"
- Custom

All may have additional text entered, eg message 4 plus 10:00 will show "Meeting until 10:00" and the text strings are localized to the system language

This feature is supported across the IP Office Small Community Network (SCN)

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Call Tagging

Feature

- Display a text message on the user's phone, Avaya one-X™ Portal for IP Office or Phone Manager, when a call is presented to it.

Benefit

- Provide additional information about the call.

Description

This feature is used to provide additional information about the call to the targeted user before they answer it. Call Tagging may be used when transferring a call from Avaya one-X™ Portal for IP Office, Phone Manager or Soft Console to give caller info if the user doing the transfer is not able to announce the call.

It is possible to add a tag to a call automatically using CTI and IP Office VoiceMail Pro. This is also possible based on an Incoming Call Route. On some telephones, displaying the Tag may mean that it is not possible to display the usual call source and target information.

Reclaim Call

Feature

- The ability to recover, or reclaim, the last call that was at your phone but is now ringing or is connected elsewhere.

Benefit

- If you just miss a call and it goes to voicemail or call coverage, you can get the call back while it is still being presented or connected through IP Office

Description

This is a special version of the Acquire Call feature that only applies to the last call at your extension.

Hunt Group Enable/Disable

Feature

- The ability for a user to enable or suspend their membership of hunt groups.

Benefit

- A user may need to temporarily join or leave individual hunt groups, for example to cover a peak of calls without changing the system programming.

Description

A team supervisor or administrator may not usually take calls for a team but at times of high traffic they may join the group to take calls and when the peak is over leave the group to resume their regular tasks. To use this feature the User must be configured as a member of the hunt group by the systems administrator, it is not possible for a user to arbitrarily join a hunt group that they have not been identified as a member of.

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Call Waiting

A User may not want people calling them to receive busy tone if they are already on another call, but have the call receive ring tone and have some kind of alert that there is a call waiting. The user can then decide to finish or hold the current call and answer the one that is waiting. The amount of information that is available about the call that is waiting depends on the type of phone the user has, or if they are using Avaya one-X™ Portal for IP Office or Phone Manager.

As Call waiting tone can be disruptive it is possible to turn the feature on or off and even suspend it for a single call – useful for conference calls.

Do Not Disturb (DND)

This is the ability to temporarily stop incoming calls ringing at a user's telephone. It will prevent the user from receiving hunt group calls and give direct callers either voicemail (if enabled) or a busy signal. This feature can be enabled/disabled from the phone or via Avaya one-X™ Portal for IP Office or Phone Manager.

It is possible to have some calls bypass the DND setting and ring the phone. For example a manager might have their secretary's extension number on the DND exceptions list. The exceptions list can be easily managed by the Avaya one-X™ Portal for IP Officeor Phone Manager. Both internal and external numbers can be on the exception list.

Dial Plan

IP Office has a very flexible numbering scheme for extensions, hunt groups and feature commands. While the system has default numbering for feature codes and extensions, they can all be re-defined. Default extensions and hunt groups have 3 digit numbers starting at 200 but these can be changed from 2 to 9 digits through the IP Office Manager. There is a default set of feature access "short codes," but these can be changed to what ever the end user requires, within limits. This is useful for example, if IP Office is replacing a system where DND was accessed by dialling *21, it is possible to change the IP Office Short Code to mimic the code of the replaced system.

In certain countries IP Office can support a Secondary Dial Tone when an access digit is dialled, though this limits some functionality like Alternate Route Selection (ARS). IP Office can also be configured to work without line access digits, by analyzing digits as they are dialled and determining if they are for an internal number or should be sent out on a line – this is valuable in SOHO installations where users will not necessarily be used to dialling an access digit for an outside line.

Paging

All Avaya digital and IP telephones supported on the IP Office that have loudspeakers can be used to receive broadcast audio messages without having to install a separate paging system. Paging can be to individual telephones or groups of telephones.

Analog extension ports can be configured for connection to external overhead paging systems, usually through an adapter, such that a port can be included in a paging group to permit mixed phone and overhead paging.

Some Avaya digital and IP telephones are able to answer a page by pressing a key while the page is going on, this terminates the page and turns it into a normal call.

This feature is supported across the IP Office Small Community Network

Intrude

The Call Intrude feature allows a user, if permission through IP Office Manager is given, to join an existing conversation whether this is an internal or external call.

A user with the "Can Intrude" option can join a call on any extension on the system, however, a User with "Cannot be Intruded" setting would prevent others from joining their call.

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Inclusion

This feature enables selected users to intrude on calls that are already in progress. The intruding party intrudes on the existing call and all parties hear a tone. The speech path is enabled between the intruding party and the called user; the other party is forced onto hold and will not hear the conversation. On completion of the intrusion the called party speech path is reconnected to the original connected party. The feature is enabled or disabled on a per user basis through the Manager.

Private Call

Users can set a status of private call using short codes or a programmed button. Private calls cannot be recorded, intruded on, bridged into or monitored.

Hot Desking

Hot Desking allows a number of users non-exclusive use the same extension. Each user logs in with their own identity so they can receive calls and can access their own Voicemail and other facilities. For example, sales personnel who visit the office infrequently can be provided with telephony and Voicemail services without being permanently assigned a physical extension. When finished, they simply log out to make the extension available to others or if users log in at another phone, they are automatically logged out of the original extension.

Remote Hot Desking

Feature

- The ability for a user to Hot Desk to other locations within the Small Community Network.
- Available on Digital, Analog and IP telephones.

Benefit

- A user can make and receive calls from any office as if using the phone on their own desk.
- Single number, improved mobility and easy access to familiar features.
- The user has access to the centralized system and personal directory as well as their call log (1400, 1600, 9600 Series and T3 telephones only).
- Great for consultants, managers, lawyers working on different offices on different days.

Description

IP Office supports remote hot desking between IP Office systems within a Small Community Network (SCN). The system on which the user configured is termed their 'home' IP Office, all other systems are 'remote' IP Offices. No additional licenses are required to support Remote Hot Desking other than the Voice Networking license on each IP500 within the SCN.

- User Settings
  When a user logs in to a remote IP Office system, all their user settings are transferred to that system.
  - The user's incoming calls are rerouted across the SCN.
  - The user's outgoing calls use the settings of the remote IP Office.
  - However some settings may become unusable or may operate differently. For example if the user uses a time profile for some features, those feature will only work if a time profile of the same name also exists on the remote IP Office.
- Break Out Dialing
  In some scenarios a hot desking user logged in at a remote system will want to dial a number using the system short codes of another system. This can be done using either short codes with the Break Out feature or a programmable button set to Break Out. This feature can be used by any user within the Small Community Network but is of significant use to remote hot deskers.

Note: Remote Hot Desking is not supported for use with Customer Contact Center (CCC) or Customer Call Reporter (CCR). Features handled by the telephone itself are not affected by Hot Desking (e.g. call log and phone speed dials).

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Relay On/Off/Pulse

IP Office is fitted with two independent switch outputs for controlling external equipment such as door entry systems. Control of these switches is via allotted handsets allowing the switches to be opened, closed or pulsed as required. Control of switches is also accessible via, Receptionist SoftConsole and VoiceMail Pro.

Pickup

Call Pickup allows a user to answer a call presented to another extension. Types of call pickup include:

- Pick up any call ringing on another extension.
- Pick up a hunt group call ringing on another extension, where the user must be a member of that hunt group.
- Pick up a ringing call at a specified extension.
- Pick up any call ringing on another extension that is a member of the hunt group specified.

This feature is supported across the IP Office Multi-Site Network (SCN).

Call Recording

Where IP Office has VoiceMail Pro installed it is possible to record a call and save the recording to the user's mailbox, a group mailbox or the voice recording library. For example, this is useful when a caller is going to give detailed information like an address or phone number and the caller will hear a warning message or tone that the call is being recorded in some countries. Where call recording is required for Quality Assurance, it is possible to set the IP Office system to automatically record a percentage of calls for later review.

Beginning with IP Office R4.0, any call (normal, conference, or intrusion) and any phone type (including IP) can be recorded. Where "advice of recording" needs to be played, IP Office will ignore Voicemail port licensing if an insufficient number of voicemail channels have been licensed.

Note: for IP telephones, a VCM channel will be required for the duration of the recording.

Telecommuter Mode

Avaya one-X™ Portal for IP Office for IP Office allows the making and receiving of calls and the retrieving of voicemails from an external phone number as if they were in the office, with one-X Portal for IP Office server providing the call control. The typical scenario is the remote worker that occasionally works from home or from a hotel room.

This feature also provides billing convenience and potential cost savings for remote workers and mobile work force as all the calls are established by IP Office: there is no need to check bills, nor to pay for expensive hotel calls.

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Twinning allows a primary extension and a secondary number (extension or external) to operate together as a single telephone. When a call is presented to the primary phone the secondary will ring. If the primary telephone does not ring, for example in Do Not Disturb, the secondary phone will not ring.

Please refer to the Mobility Chapter for more details.

Internal Twinning

This is typically used in scenarios like workshops or warehouses where team supervisors may have a desk with a fixed phone but also have a wireless extension (e.g. DECT). When a call is made from either twinned phone, the call will appear to have come from the primary phone. Other users of the system need not know that the supervisor has two different telephones. The supervisor's Coverage Timer and No Answer Time are started for the call and if the call is not answered within that time, the call will be delivered to available coverage buttons (if applicable) and then Voicemail (if applicable).

The following features are supported with internal twinning

- Follow Me To
- Follow Me Here
- Forwarding
- Do Not Disturb (inc exceptions)
- Context less hunt group actions: Membership / Service Status / Fallback Group configuration
- Voicemail On / Off / Access
- Call Log (Central Call Log for T3 and 1600 telephones only)
- Redial (Central Call Log for T3 and 1600 telephones only)
- Personal Directory Entries (for T3 and 1600 telephones only)

Mobility Features include:

- Mobile (external) Twinning
- Mobile Call Control
- Mobility Callback
- Avaya one-X Mobile Client Support

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Key and Lamp Operation

IP Office offers a full range of Key and Lamp features on Avaya feature telephones. These features include; Line Appearance, Call Appearance, Bridged Appearance and Call Coverage. As these features require a phone with buttons and indicators, the features are only supported on certain Avaya digital and IP telephones. Key and Lamp operation is not supported on analog telephones.

IP Office can have a ring delay set on each appearance button to allow time for the target number to answer before other extensions ring, or visual alert only without ring.

In Key and Lamp operation, IP Office supports up to 10 buttons on each telephone and 10 telephones with the same line appearance.

Appearance Buttons

Feature

- Use the programmable buttons available on Avaya digital and IP telephones to represent individual calls.
- Answer, originate and join calls by pressing the appropriate appearance buttons.

Benefits

- Indication of calls connected and calls waiting.
- Handling of multiple calls from a single phone.

Description
Many Avaya digital and IP telephones supported by IP Office have programmable buttons. These buttons can be assigned to appearance functions that allow the handling of calls. These functions are:

- Line Appearance Buttons

  Used to indicate make and answer calls on a specific external trunk.

- Call Appearance Buttons

  Used to handle multiple incoming and outgoing calls from a user's extension.

- Bridged Appearance Buttons

  Used to match the call appearance buttons on a colleagues extension.

- Call Coverage Buttons

  Used to indicate unanswered calls ringing at a colleagues extension.

Line Appearance

A Line Appearance is a representation of a trunk line on the IP Office system where the indicator tracks the activity on the Line. Only external calls can be answered or made on Line Appearances. Line appearances can be used with Analog, E1 PRI, T1 PRI and BRI trunks PSTN trunks. They cannot be used with E1R2, QSIG and IP trunks.

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Call Appearance Buttons

Feature

- Uses a programmable button on the Avaya digital and IP telephone to represent an incoming or outgoing call.
- Separate buttons are used to represent each simultaneous call that the user can make or answer.
- Where possible, the status of the calls (ringing, connected or held) is indicated by the button indicator.

Benefit

- Call appearances allow a single user to make, answer and switch between multiple calls by pressing the appropriate call appearance button for each call.

Description

On Avaya IP Office digital and IP telephones that have programmable buttons, those buttons can be set as call appearance buttons through the IP Office Manager. The number of call appearance buttons set for a user determines the number of simultaneous calls they can make and answer.

Note that the use of call appearance buttons overrides IP Office call waiting features. It is only when all call appearances are in use that subsequent callers receive either busy tone, voicemail or follow a forward on busy action

When call appearance buttons are used, a minimum of three call appearance buttons is recommended where possible, although some telephones are restricted to two call appearance buttons by the number or design of their programmable buttons.

Alerting/Ring Tone for Covered Calls

Feature

- The alert signal (ring tone) for incoming calls for covered telephones can be set to the following values: Ring (default), Abbreviated Ring and No Ring.

Benefits

- The ability for a user to choose how the covered call will alert.
- To keep the alerting noise low in open-plan offices.

Bridged Appearance Buttons

Feature

- Allow the user to have an appearance button that matches another user's call appearance button.

Benefit

- Answer and make calls on behalf of the other user.
- Audible indication of calls presented to the bridged user, where programmed
- Visual indication of when the other user has calls presented, held or connected.
- Join and exchange calls using the paired call appearance and bridged appearance buttons.

Description

A bridged appearance button matches the activity on one of another user's call appearance button. For example, when the call appearance shows a ringing call, the bridged appearance button will also show the ringing call and can be used to answer that call.

Similarly, if the bridged appearance button is used to make a call, the call activity is shown on the matching call appearance button. The call appearance button user can join or takeover the call using their call appearance button.

Bridged appearance buttons allow paired 'manager/secretary' style operation between two users, and are only supported for users who have call appearance buttons.

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Call Coverage

Feature

- Allow unanswered calls to alert at other user extensions and be answered there before being forwarded or going to voicemail.

Benefit

- Provide users the opportunity to answer colleague's unanswered calls before they go to voicemail.

Description
When a user has an unanswered call ringing, after a configurable delay, the call will also start alerting on any call coverage buttons associated with the user on other extensions. The call can then be answered by pressing the call coverage button. If still unanswered the call is forward or goes to voicemail as normal.

The time a call rings before also alerting on any associated call coverage buttons can be adjusted for each user.

External Call Lamp Indication

Feature

- New system wide configuration parameter to select the LED flash pattern for external calls on Bridged and Coverage appearances.

Benefit

- The user can determine if the covered call is internal or external.

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Outbound Call Handling Features

Every business needs to make calls, but depending on the type of business these calls may need to be treated in a special way, such as recorded against a project or client through the use of Account Codes. A business may have several sites linked via a private network but certain users, like customer services agents, may need to be able to call colleagues in other offices even when the network is busy, while other users can wait for a line to come free, Least Cost Routes can automatically translate the internal number to a direct dial call over the public network while other users wait.

Account Codes

Feature

- Associate an account code with a call.
- Validate account codes used against list stored by the IP Office.
- Include the account code used with call log details.

Benefit

- Through the call records, group calls by account code for the purpose of call costing and tracking.
- Restrict outgoing calls by requiring users to enter a valid account code.

Description

IP Office stores a list of valid account code numbers. When making a call or during the call, the user can enter the account code they want associated with that call. IP Office will check the account code against its list of valid codes and request the user to re-enter the code if it is not valid. For incoming calls, the Caller ID can be used to match it with an account code from the IP Office's list of valid codes and report the account code with the call for billing.

Individual users can be set to Forced Account Code operation where they are required to enter a valid account code before making external calls. By using IP Office Short Codes it is possible to identify certain numbers or call types as requiring a valid account code before permitting the call to proceed, for example long distance or international numbers. Analog phone users can only enter account codes before making a call or in response to an audible system prompt to enter a code when making the call.

Account codes can also be entered through the Avaya one-X™ Portal for IP Office and Phone Manager. A system wide setting determines whether Phone Manager will display a list of account codes from which the user can select the code required or will hide the account code list.

In all the cases above, the account code entered is included with the call details in the IP Office's call record output. (SMDR).

Authorization Codes

Authorization codes allow an IP Office user to go to another extension on the system and make calls using their personal toll restrictions; this may grant the user greater or fewer privileges than the normal owner of the extension they use. Since Authorization Codes are independent of Account Codes, the user has to enter both if the required by the system configuration. All entered codes are logged in SMDR.

Dial Emergency

Dial emergency is an IP Office Short Code and, permits certain numbers to be dialed regardless of call barring or a phone being logged out.

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Call Barring

Feature

- It is possible to prevent or allow calls to certain numbers such as international numbers or premium rate numbers for individual users or on a system wide basis.

Benefit

- Restrict the dialing of specific numbers or types of numbers system wide.
- Restrict certain users from dialing specific numbers or types of numbers.

Description
IP Office supports call barring at many levels. Short codes can be used at the system or individual user level to block the external routing of specific numbers or types of numbers. Typically the barring short codes are set to return busy tone, however they could route the call to an alternate number or to a Voicemail service that returns a 'barred dialing message'.

For users, the short codes can be allocated to a User Rights template. This template is then applied to the Users whose calls need restriction. In addition to barring the dialling of certain numbers, IP Office can be set to bar the forwarding of calls to external numbers on a per user basis.

Alternate Route Selection (ARS)

IP Office supports Alternate Route Selection(ARS). If a primary trunk is unavailable, then ARS provides automatic fallback to an available trunk (e.g., analog trunk fallback if a T1 or SIP trunk fails, or use PSTN for SCN fallback).

By configuring ARS, calls may be routed via the optimum carrier. Time profiles can also be used to allow customers to take advantage of cheaper rates or better quality at specific times of day.

Multiple carriers are supported. For example, local calls are to go through one carrier between specific hours and international calls through an alternative carrier. Carrier selection using 2-stage call set up via in-band DTMF is possible. It is possible to assign specific routes on a per user basis, e.g. only allow expensive routes to be used by critical staff.

Maximum Call Length

This feature allows the system to control the maximum duration of any call based on the dialed number. This could be used for controlling calls to cellular networks or data calls made over the public network to ISPs.

Transferable Dial Out Privilege

Feature

- A privileged user (Operator / Supervisor) can transfer an outside line (secondary dial tone) to a user that does not have external privileges.

Benefits

- Provides the company with control over external calls
- Provides the ability to provide outside line access for restricted telephones . (e.g., those in public areas or conference rooms)

Idle Line Preference

Feature

- Going off hook will select the first Idle line appearance and the user will be connected to an outside line.

Benefit

- Provides the ability to select a specific external line (for companies that prefer to work in key-system mode)

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Forwarding

This is the ability to forward a user's calls to another extension or external number such as a Mobile/Cell Phone. Calls can be forwarded in a number of ways and if the call is not answered at the forward destination it will go to IP Office voicemail if enabled for the user and call supervision is available. Once the numbers have been entered, the user can toggle the forwarding to be active or not as required without having to re-enter the numbers.

If the user is a member of a hunt group, some types of hunt group calls can also follow forward unconditional. Users can select if forwarding is applied to external calls only, or all calls. Call forwarding is processed after Do Not Disturb and Follow-Me conditions are tested.

Associated Features

- Do Not Disturb (DND)
- Voice Mail (VM)
- Follow Me
- Hunt groups
- No Answer Interval

Precedence

- Forward Unconditional
- Forward Busy
- Forward No Answer

Coverage to Operator

Feature

- An operator or a group of operators can be configured to provide coverage for external calls that would otherwise go to voicemail.

Benefit

- Any unanswered calls will be routed to an operator or a Group of operators. For example, local government offices prefer to provide a personal service (rather than voicemail).

Forward on Busy

If enabled, this forward will be triggered when the user is busy and another call is routed to them, but does not include calls for a hunt group that they may be a member of. A user is normally considered to be busy when they are on a call but depending on call waiting settings and key and lamp features this may not be the case.

Forward on No Answer

This forward is triggered if a call has been ringing for a user but they haven't answered it within the configured answer time, this includes calls that have been indicating call waiting if enabled.

Forward Unconditional

This sends all calls for the user to the forward unconditional number, but if the call is not answered within a user's timeout period the call will be sent to IP Office voicemail, if enabled.

Unconditional Forward to Voicemail

Feature

- The ability to forward calls to Voicemail, even when the user's voicemail is not activated.

Benefit

- This feature allows a user who is familiar with the existing IP Office functionality of diverting all calls to a person or a group to follow the same pattern and divert all calls to a voicemail box.

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Forward Hunt Group

Calls for a hunt group that the user belongs to can also follow forward unconditional. The hunt group must be set for either sequential or rotary ring type and if the call is not answered at the forward destination it will follow the hunt group call handling instead of going to voicemail. This can be particularly useful in a sales or support environments where a number of people may be out of the office on Mobile/Cell telephones and still participate in the hunt group as if in the office.

Follow Me

Follow-Me gives mostly the same behaviours as the user's home phone (Paging, hands free answer, etc) whereas Forwarding doesn't. Follow-Me is typically used when a user is going to be working away from their desk, for example in a workshop. All the call settings the user has on their main phone will apply to calls that follow the follow-me feature, including forward on busy or no answer.

Follow-Me can be set either from the users main phone – Follow-Me To – or from the phone where they want calls to be received – Follow-Me Here. Several people can have their telephones forwarded to a follow-me destination and if the phone has a display it will indicate who the call is for.

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Programmable Buttons

As well as the usual dialing keys, Avaya digital and IP telephones have dedicated function buttons like Mute, Volume, Hold, Conference and Transfer. In addition to these, on many Avaya digital and IP telephones there are keys that can be programmed with a range of selected special functions. These keys can be used for calling other extensions on the system (Direct Station Select or DSS keys), or can be used for options from speed dialing numbers to controlling features such as Do Not Disturb. Many features use an indicator to show whether a feature is enabled. Button programming is done through the IP Office Manager as part of the system configuration, although some telephones allow the user to program buttons and functions where given administration rights.

Busy Lamp Field (BLF) Indicators

Feature

- Status indicators which show the status of a programmable buttons associated feature or function.

Benefit

- Indication of when a button or associated feature is active.

Description

Avaya digital and IP telephones have programmable buttons which can be assigned to various features. When those buttons include some form of BLF indicator, the button can also be used to indicate when the feature is active. For example, a button associated with another user will indicate when that user is active on a call. A button associated with a group will indicate when the group has calls waiting to be answered.

The directory entries in Avaya one-X™ Portal for IP Office and the speed dial icons within the IP Office Phone Manager and SoftConsole applications also act as BLF's. When the icons are associated with internal users, the icons will change to indicate the current status of the users.

Avaya one-X™ Portal for IP Office shows these conditions:

State	Icon	Description
Available		The normal state for a user showing that their work extension is not in use.
Busy		The normal state for a user showing that their work extension is currently on a call.
Do Not Disturb		The user has set Do Not Disturb. Calls to them will go to voicemail if enabled or else get busy tone unless you are in the user's Do Not Disturb exception list.
Logged Out		The user has logged out from their phone. Calls to them will most likely go to voicemail if available.
Other		This icon is used when the status is not known.

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Call History

Feature

- Access to a list of called, received and missed calls from the phone and/or IP Office user productivity applications.

Description

IP Office keeps a record of calls made and received, including unanswered calls. Details are store for both users (maximum 30 entries) and hunt groups (maximum 10 entries). The method of operation varies according to the phone type but in all cases the call records can be used for return calls.

With IP Office Release 6, a centralized call log is supported by 1400, 1600 and 9600 Series telephones (not 1403 or 1603/1603SW) and Avaya one-X™ Portal for IP Office (leading to consistency between the two). 5400 Series and 5600 Series call logs are stored on the phone.

Avaya one-X™ Portal for IP Office accesses the centralized call log and therefore presents a record of all calls even if the user's PC is switched off when a call is received. Call log entries can be added to the personal directory..

one-X Portal for IP Office can display call histories for all calls, missed calls, inbound calls and outbound calls. Entries in the call history can be used for return calls, sorted and added to the one-X Portal local directory or speed dials.

Benefit

- Missed calls are easily tracked and can be returned.
- Call log data is retained even after power down and a system reset (1400, 1600, 9600, T3 and Avaya one-X™ Portal for IP Office).
- Centralized call log is supported in the SCN when using hotdesking (1400, 1600, 9600, T3 and Avaya one-X™ Portal for IP Office).
- Consistency between desktop telephones and user productivity application (1400, 1600, 9600, T3 and Avaya one-X™ Portal for IP Office).

Language

Avaya digital and IP phone menus and displays are available in many languages and usually the system default setting will be applicable to all telephones, however it is possible to have language set on an extension by extension basis, this will also change the language of menus for IP Office Voicemail.

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Centralized Personal Directory

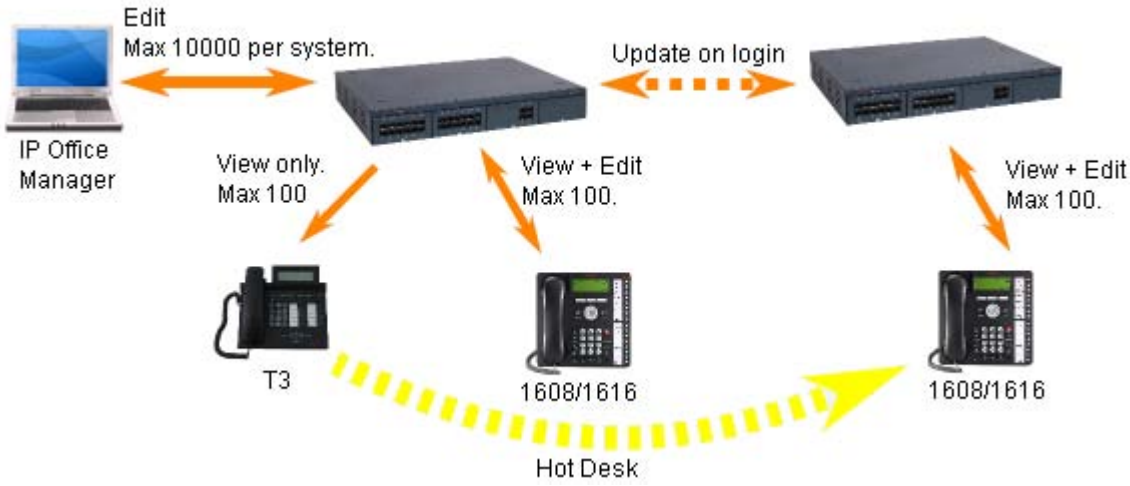
Feature

The IP Office Personal Directory is a list of up to 100 numbers and associated names stored centrally in the system for a specific user. A directory entry can be used to label an incoming call on a caller display telephone or on a PC application. The directory also gives a system wide list of frequently used numbers for speed dialing via Avaya one-X™ Portal for IP Office, Phone Manager or a feature phone with a suitable display.

For example "Mr Smith" can be displayed when a known Caller ID is received. A user can also select "Mr Smith" in the Directory List in Phone Manager, Avaya one-X™ Portal for IP Office or on a display phone to speed dial this number.

Description

- The personal directory capability allows up to 100 entries per telephony user.
- The system limit is:
  - 10800 for IP Office 500 and IP Office 500V2.
  - 3600 for IP Office 412.
  - 1900 for IP Office 406 V2.
- All entries may be added, deleted or modified by Manager, Avaya one-X™ Portal for IP Office, a telephone, or an external service.
- The personal directory data is sent/updated whenever the user is logged in a SCN.



Benefit

- The user has a personal directory independent from the phone (1400, 1600, 9600 and T3 only) at which they are is logged in
- The personal directory is always synchronized between telephone and Avaya one-X™ Portal for IP Office .

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Centralized System Directory

Feature

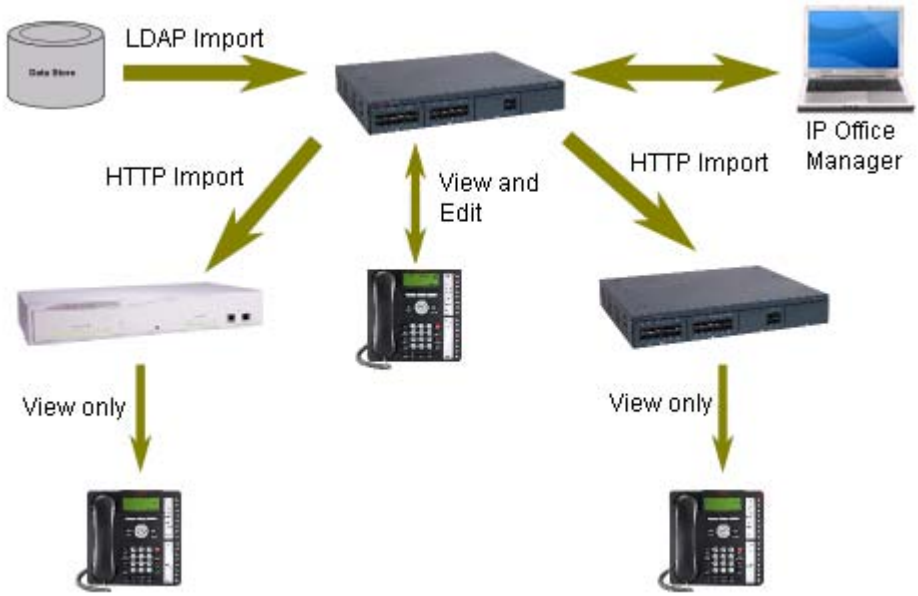
The IP Office Directory is a list of up to 5000 numbers and associated names stored centrally in the system. A directory entry can be used to label an incoming call on a caller display telephone or on a PC application. The Centralized System Directory also gives a system wide list of frequently used numbers for speed dialing via Avaya one-X™ Portal for IP Office, Phone Manager or a feature phone with a suitable display.

For example "Head Office" can be displayed when a known Caller ID is received. A user can also select "Head Office" in Avaya one-X™ Portal for IP Office, or the Directory List in Phone Manager, or on the display phone Directory to speed dial this number.

Description

- External directory entries can be obtained from up to three different sources for each IP Office:
  - Locally stored in the IP Office configuration. Created and edited using IP Office Manager or by selected 1600, T3 or Avaya one-X™ Portal for IP Office users.
  - Temporary entries imported at regular intervals from an LDAP server.
  - Temporary entries imported at regular intervals by HTTP from another IP Office. This allows the directory entries on one IP Office to be shared by other IP Offices as a centralized system directory for an SCN.
- The total maximum number of entries is 5000 with entries in the IP Office configuration taking priority over temporary imported entries.

	Manager	LDAP	HTTP	Overall Maximum
IP500 IP500 V2	2500	5000	5000	5000



Benefit

- Ability to use external directory services with different interfaces
- One system directory for a Multi-Site Network (SCN)

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Self-Administration

The IP Office administrator may give select users the ability to change some of the phone settings themselves. For example, button programming. The range of changes that the user can make depends on the phone type in use.

On Hook Dialing

Avaya digital and IP telephones allow the user to make calls by just dialing the number on the keypad, without having to lift the handset or pressing a speaker button. Usually the call progress can be monitored using the speaker in the phone, on telephones that support handsfree the whole conversation can be had without having to lift the handset.

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Inbound Call Handling

IP Office offers several features to provide versatile inbound call processing, including PC based applications, and a standards-based TAPI interface for 3rd party applications.

Incoming Call Routing

Incoming calls can to be presented to an Operator who then decides where to pass the call, but IP Office supports intelligent call routing capable of making routing decisions based on a number of criteria.

The system currently supports routing based on;

- Call presentation digits from the exchange such as DDI/DID or ISDN MSN.
- Calling telephone number or Caller ID (This could even be part of the number received such as an area code).
- ISDN sub-address.
- ISDN/PRI service type i.e. Voice Call, Data Call, etc.

It is even possible to look for multiple criteria so, for instance, a DDI/DID call to a sales group could be handled differently depending on which part of the country the call is originating from.

Each incoming Call Route also supports a secondary destination 'Night Service' that can provide alternative routing for an incoming call based on 'time of day' and 'day of week' criteria, as well as calendar-based routing for specific dates.

Calls that cannot be routed to the configured destination are re-routed to a user defined 'Fall Back' destination. This can be particularly useful where calls are normally answered by an auto-attendant and a network fault occurs.

Where multiple call routes are set up to the same destination, a Priority level can be associated with the call. This priority level is used to determine a call's queue position in place of simple arrival time, but note that calls already ringing a free extension are not considered queuing and are not affected by a high priority call joining a queue (unless the option" Assign Call On Agent Answer" is selected for that hunt group). A Priority Promotion Timer can be configured to increase the priority of calls which have been in the queue for more than a defined time.

An optional tag can be added to calls on the Incoming Call Route, which can be displayed on the alerting telephone.

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Hunt Groups

A hunt group is a collection of users, typically users handling similar types of calls, e.g. a sales department. An incoming caller wishing to speak to Sales can ring one number but the call can be answered by any number of extensions that are members of the hunt group.

Four modes of call presentation are supported on IP Office;

- Sequential
  One extension at a time sequentially always starting at the top of the list.
- Collective
  All extensions in the hunt group simultaneously.
- Rotary
  Start with the extension in the list immediately following the extension that answered the last hunt group call.
- Longest Waiting
  Start with the extension that has been free for the longest time.

An option has been added, "Assign Call On Agent Answer", which means that CTI applications will always correctly report the details for the call that is alerting. The "Assign Call On Agent Answer" and ensures that the call at the head of the queue is always answered first.

If all extensions in the hunt group are busy or not answered, another hunt group, called an Overflow Group, can be used to take the calls. An overflow time can be set to stipulate how long a call will queue before being passed to the Overflow Group (this can be configured either for individual calls or for all calls in the group). The system can change the status of users who do not answer a hunt group call presented to them. The user can be put into busy wrap-up, busy not available or logged out. The change of status can be set per user and the use of this option can be set per hunt group.

Outside normal operation a hunt group can be put into two special modes; Night Service and Out of service.

In Night Service calls are presented to a Night Service Group. This can be controlled automatically by setting a time profile which defines the hours of operation of the main group or manually using a phone feature code.

Night service fallback using a time profile is no longer applied to a hunt group already set to Out of Service.

The Out of Service mode is controlled manually from a phone. While in this mode calls are presented to the Out of Service group

Voicemail can also be used in conjunction with hunt groups to take all group related messages, play an announcement when the hunt group is in Night Service or Out of Service mode and give announcements while a call is held in a queue. For internal voicemail use a broadcast option is provided. This feature will alter the voicemail box operation so that the message notification will only be turned off for each hunt group member when they retrieve their own copy of the message.

Multi-site Networking/Small Community Networking (SCN) Distributed Hunt Groups

Hunt groups in a Small Community Network can include members located on other systems within the network.

Note: Distributed Hunt Groups are not supported for use with CCC or CCR.

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Night Service

When a hunt group is in Night Service mode, the hunt group is temporarily disabled. Callers to this hunt group will:

- Pass to a Night Service Fallback group used to provide cover, e.g. pass calls to a manned extension or an external number
- Be played the Out of Hours greeting if voicemail is operational
- Receive the busy tone

A hunt group can be switched in or out of Night Service mode by a user dialing the appropriate short code – by any extension or by specific users.

Time Profiles

Time Profiles can be used to define when a Service, Hunt Group, Least Cost Route, Conference Bridge or a user's dial-in facility are operational. For example, a time profile can be used to route hunt group calls to a manned extension or voicemail outside of office hours, or be used to apply different Least Cost Routes at varying times of day to take advantage of cheaper call rates. Multiple Time Entries can be created so that a Time Profile can be used to define specific hours in the day e.g. 09:00-12:00 and 13:00-17:00. Outside of a Time Profile, voice calls would be re-routed according to the configuration but any currently connected calls at the time the Time Profile changes would not get cut off as the change only affects the routing. Data calls will get cut off as the time profile goes out of service but a new data call will start immediately if specified. Time Profiles can also be based on specific calendar dates to make allowance for public holidays or other events.

Queuing

Queuing allows calls to a hunt group to be held in a queue when all extensions in the group extension List are busy. When an extension becomes free the queued call is then presented. The definition of queued calls now includes ringing calls and calls waiting to be presented for ringing. The queue limit can be set to control the maximum number of calls to wait against a hunt group.

While queuing, if Voicemail is operational, the caller will be played the announcements for this hunt group.

Announcements

Hunt group announcements are separated from hunt group queuing and can be used even when queuing is off. Hunt group announcements are now supported by Embedded Voicemail in addition to VoiceMail Pro.

Further, times for the first announcement, second announcement, and between repeated announcements are configurable.

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Contact Center Features

Contact Centers have specific needs for reporting on how calls are handled and these are covered in a separate section of the Product Description. Basic handling of telephony requirements for a Call Center is a standard part of IP Office from Automatic Call Distribution (ACD) and Call Queuing to agents logging on and selecting the groups that they service.

Login

A contact center agent function, login is required before the agent is able to make or receive calls from their phone. A login idle period can be specified which will dictate how long an extension can be idle before the user is automatically logged out, ensuring that an extension is not left logged in and calls go unanswered.

Monitor Calls

A user can monitor other peoples' calls by listening in. This feature is not available by default; it must be specifically enabled in the system configuration. An option exists to have a beep tone indicate when monitoring is in use. The user is only able to listen; they cannot speak into the conversation being monitored.

Acquire Call

Feature

- Takeover a call currently connected at another extension. This feature is also known as "Call Steal".

Benefit

- Assist a colleague who indicates they want you to take the call.

Description

The Acquire Call function can be setup as a special short code or programmed against a button on an Avaya digital or IP phone with programmable buttons. Use of the feature is subject to IP Office intrusion control settings, the user acquiring the call must be set to be able to intrude and the user whose call is being acquired must be set to can be intruded. Acquire call works in two ways, invoked with or without a number:

Without a value in the number field

- This allows a user to reclaim a call that was ringing on their phone but has now gone elsewhere, for example to Voicemail or Forward No Answer destination. The Intrude settings are not checked and the call can be reclaimed even if it has been answered.
- If the last call to ring this User is no longer ringing or connected on the system, the feature will fail.

With a number, where the number is the telephone number of a user who currently has the call to be acquired.

- If the user has a call ringing or waiting Acquire Call will act like the Call PickUp Extension short code and the user executing Acquire Call will be connected to the oldest ringing/waiting call.
- If the User has a connected call with no call waiting and the Intrude settings of the two Users allow it, the call will be connected to the user executing the Acquire Call and the other user will be disconnected.
- If the User does not have a call the feature will fail.

Queue Threshold Alert

When the number of calls queued against a hunt group exceed a threshold, the system can be configured to alert at a selected analog extension port. Typically the User to Alert will be a loud ringer or other alerting device. The alert does not present a real call.

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Miscellaneous Features

Conference Calls

Calls can be placed on hold and a conference created using either the phone or desktop applications. Additional conference members may be added, however a single conference may not have more than 64 members.

For the IP500 and IP500 V2 the total conferencing capacity is 128 channels. This could be alternatively 2 conference of 64 users, 3 conferences of 42 users or any other combination (maximum 64 participants per conference call).

Only two calls connecting through analog trunks are permitted in any single conference.

For more information on managing conference calls, refer to the Audio Conferencing chapter 334.

Dial On Pickup

Also known as "Hotline". Automatically dials a specified extension when the phone is taken off hook. This facility is commonly used in unmanned reception areas or for door entry systems to allow visitors to easily gain assistance.

Off Hook Operation

Off-Hook Station is designed for users who want their analog phone to operate like digital or IP feature phone, to isolate the user's phone idle state from the Hook state. This is a useful feature when using Avaya one-X™ Portal for IP Office, Phone Manager or SoftConsole to control the phone state when using a headset on an analog telephone and with call control and dialing from Avaya one-X™ Portal for IP Office, or SoftConsole.

E911 / Emergency Call

This is a specific service for North America. When an emergency call is connected, IP Office provides calling party information to an external line interface unit. The external unit carries out a number to text translation and forwards this to the emergency services bureau so that the originating location of the call is clearly identified.

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System Short Codes

Short codes are used as commands the IP Office to make changes for the user, group or system, so need to set up with consideration to security. The command may need additional information included with it, such as for forward, the phone number forwarded to. Short codes are a flexible and quick way of setting up certain features. IP Office has short codes provided by default on the system, or more advanced codes that need programming by the system administrator.

Examples

The method of detailing a short codes settings lists the short code fields separated by a /.

*17/?U/VoicemailCollect
A user dialing *17 is connected to voicemail.
*14*N#/N/FollowMeTo
If a user dials *14*210# at their own extension, their calls are redirected to extension 210.

Dialing Short Codes

The following types of short code applied to on-switch dialing. The result may be an action to be performed by the IP Office, a change to the user's settings or a number to be dialed. The order below is the order of priority in which they are used when applied to user dialing.

User Short Codes

These are useable by the specific user only.

User Rights/User Restrictions Short Codes

These are useable by any users associated with the user rights or restrictions in which they are set. They can be overridden by individual user short codes.

User and User Rights short codes are only applied to numbers dialed by that user. For example they are not applied to calls forwarded via the user.

System Short Codes

These are available to all users on the IP Office system. They can be overridden by user or user rights short codes.

Post-Dialing Short Codes

When any the short code above result in a number to be dialed, further short code can be applied to that number to be dialed. This is done using the following types of short codes.

ARS (Alternate Route Selection) Short Codes

The short code that matches dialing can specify that the resulting number should be passed to an ARS form. The ARS form can specify which routes should be used for the call by using further short code matches and also provide option to use other ARS forms based on other factors such as time and availability of routes.

Transit Network Selection (TNS) Short Codes

Used on T1 ISDN trunks set to use AT&T as the Provider. Applied to the digits presented following any other short code processing.

Incoming Number Short Codes

On certain types of trunks short codes can be applied to the incoming digits received with calls.

Line Short Codes

These short codes are used to translate incoming digits received with calls. The stage at which they are applied varies between different line types and may be overridden by an extension number match

The full set of short code commands are listed in the IP Office Manager, refer to product configuration documents for more detail on how to set them up.

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Chapter 7.
IP Telephony

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7. IP Telephony

Technological innovation is changing the way we communicate. This time it is coming in the form of changing the way telephone calls are transmitted. It brings with it several new capabilities that change the meaning of the phrase telephone call through the use of Voice over Internet Protocol (VoIP). Basically, VoIP means "voice transmitted over a packet data network." VoIP is often referred to as IP Telephony because it uses the IP protocols to make possible enhanced voice communications throughout the world, wherever IP connections have been delivered. IP Telephony unites a company's many locations- including mobile workers- into a single converged communications network. Telephony calls using VoIP go above and beyond what's been possible in the past. When it comes to placing telephone calls, VoIP provides a range of support services and features unequalled in the world of telephony, but above all deliver them at low cost.

How Does VoIP Work?

Voice over Internet Protocol means basically what the acronym states: Voice travels over an Internet Protocol. Internet Protocol refers to the type of rules that the network uses to send and receive signals. IP Telephony works by converting voice communications into data packets. Conveniently, it runs on the popular Ethernet LAN (local area network) technology, which currently supports over 96 percent of the worlds companies' LANs.

Circuit-switched or Time-Division Multiplexed Telephony

Before digital networking with the Internet took off, everyone had to use the "Plain Old Telephone Services" (POTS). These run over a network called the Public Switched Telephone Network (PSTN). The PSTN has been around since the telephone was invented in either analog or digital form using circuit switched technology where the telephone call gets exclusive bi-directional use of a wire – or circuit – while the call is in progress. Because the circuit is exclusive to each conversation, PSTN and private branch exchanges (PBXs) must be sized to cope with peak demand and have enough circuits available for all expected conversations. This is not a flexible approach and results in a lot of infrastructure investment that the telephone companies need to recoup, via the cost of access charges and calls. The Internet has changed this – where data services have driven down access charges and allowed voice to "travel for free" over a multipurpose data network.

Packet-Switched Telephony

Unlike circuit-switched connections, which always require use of dedicated bi-directional circuit for the duration of a call, VoIP technology has enabled telephony and other new and novel features and services to run over fixed and wireless networks including private local area networks. These newer network types use packet-switched protocols. Packet-switched VoIP puts voice signals into packets. Along with the voice signals, VoIP packets include both the sender's and receiver's network addresses. VoIP packets can traverse any VoIP-compatible network. Along the way, they can choose alternate, shared paths because the destination address is included in the packet. The routing of the packets is not dependent on any particular network route which means the network provides can provide a reliable service at a fraction of the cost of circuit switched providers.

What Advantage Does IP Office Have?

IP Office can provide support of PSTN, SIP, POTs, digital time division multiplexed telephones AND digital IP telephones all on the same system. This means you don't have to abandon the past to embrace the future, IP Office allows all the technologies to co-exist. IP Office connects to the PSTN and to IP trunks (the VoIP equivalent) so providing a "Hybrid" PBX function – where both legacy and future technologies can be used together to minimize operating costs and offer optimize business communications through both voice and data.

IP Office has digital telephones built on both TDM and IP technology that provide the same user interface offering a flexible choice of solution that can mix, for example TDM telephones in the office and IP telephones at a remote site of at home. With the choice of IP telephones including real and virtual (software) telephones, IP Office can take communications to a new level.

Buying IP Office allows you choice – you can use the pure POTs or the pure VoIP capabilities of IP Office, or use both at the same time to allow seamless technology transition of your business without the disruption of having to choose between them now.

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IP Office Turns VoIP into IP Telephony

In order to make use of VoIP, IP Office uses signaling protocols called H.323 , and Session Initiation Protocol (SIP) which allow IP Office to establish end-to-end connections for the voice path through the IP network. It ensures each end of the connection is able to transmit and receive voice and provides the network addressing for end to end packet transmission. IP Office also allows for connecting between the different technologies by translating the signals they use, for example an analog telephone may wish to connect to a VoIP destination. This requires both the signaling and voice transmission to be translated – IP Office does this easily as it contains technology elements called gateways and gatekeepers that enable translations to happen.

With a conventional telephone system you plug your analog or digital TDM telephone into an extension socket connected to your PBX or Key System. With IP Telephony you connect your digital IP telephone to your IP PBX via the LAN. There are two basic types of IP telephones:

- A physical telephone, which looks very similar to a standard telephone (IP Hard Phone)
- A software application (IP Office Video Softphone) which runs on the user's PC, allowing them to use either a headset/microphone to make/receive calls anywhere they have IP connection

IP telephony has the advantage of allowing extensions to be deployed both locally and remotely through the use of IP routing and IP VPN services.

When making use of IP telephony, there are a number of data centric considerations such as which data types have priority on the IP network when there is contention. This is set with IP/TCP "quality of service" and should not be ignored. In situations where LAN Bandwidth is limited, a quality of service capable LAN switch should be used to ensure voice packets are transmitted with the required priority on the network. If not, the conversation carried over IP appears as broken up (due to delays) or has unacceptable delays introduced in the conversation (latency and jitter). With IP hardphones there is need for Power over Ethernet (PoE), or local phone power supplies to be provided to the telephones as the IP telephones are no longer powered by IP Office – a list of Avaya approved PoE options is available at the end of this section.

Gateways, Gatekeepers and H.323 - Technology Overview

IP Office uses the H.323 signaling protocol which has the following architectural components

- Telephones are H.323 service endpoint devices that can support Audio calls. Other types of H.323 devices can support video as part of H.323
- Gateways provide media translation to allow calls to be made to non-H.323 devices, for instance an analog telephone or the public network to connect with a H.323 device
- Gatekeepers control the call processing and security for H.323 devices
- Multipoint Connection Units (MCU) for conferences by adding together media streams

These elements are grouped together in what is known as an H.323 zone (a zone is analogous to a PABX). Each zone has a single Gatekeeper that can be considered as the brains of the system dealing with call distribution, call control and the management of resources. On power-up, IP telephones, Gateways and MCU make registration requests to a Gatekeeper which then authenticates (accepts or rejects) their request to become a member of the zone. Once accepted, a telephone wishing to make a call sends a call set-up message to the Gatekeeper which then determines how to route the call and will then send an alert to the called telephone, or if the call is to a non-H.323 telephone establish the call via a Gateway within the zone.

The design of IP Telephony systems has been driven by open standards. Digital IP telephones, Gateways and Gatekeepers all support the H.323 standard and it is this that allows devices from different manufacturers to work together. IP Office has an optional integral Gateway (Voice Compression Modules) and Gatekeeper functionality required to provide a fully functional IP Telephony solution.

SIP – Session Initiation Protocol

IP Office supports SIP, which is a signaling protocol used for creating session oriented connections between two or more locations in an IP network. SIP trunking is a service offered by and ITSP (Internet Telephony Service Provider) that permits businesses with an IP Office installed to use Voice-over-IP (VoIP) outside the enterprise network by using the same connection as in Internet connection.

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- Gatekeeper
  The IP Office gatekeeper allows the registration of up to 384 IP extensions on the IP500, less the number of analog and digital TDM telephones already configured on the system.
- Gateway
  The Voice Compression Module provides the H.323 gateway function that allows IP extensions to make calls to other non-IP devices. The maximum number of simultaneous calls is limited by the number of channels available on the Voice Compression Module. IP Office must be fitted with an optional Voice Compression Module to enable IP telephony.
- Silence Suppression
  Silence suppression is a technique used to make the best use of available bandwidth, such as the connection over which the caller is listening, not speaking. Silence suppression works by sending descriptions of the background noise, rather than the actual noise itself, during gaps in conversation thereby reducing the number and frequency of voice packets sent on the network. Background noise is very important during a telephone call. Without noise the call will feel very unnatural and give a perception of poor quality.
- Compression
  IP Office supports a wide range of voice compression standards including G.711, G.729a and G.723.1. The method of compression can be either automatically established on a call-by-call basis or be configured on an individual extension basis.
- Fast Start
  When fast start is supported by an IP extension, this facility reduces the protocol overhead allowing an audio path to be established more quickly.
- Out of Band DTMF
  When out of Band DTMF is configured on an IP extension, the extension will signal to the other end of the connection which digits need to be regenerated by a local DTMF generator on behalf of the sending IP extension. This is useful when navigating external voicemail systems and Auto-Attendants.
- Direct Media Path
  Direct Media Path allows the speech path between two IP extensions (after call setup) to be routed directly to each other. This allows the IP Office system to free up voice compression resources after establishing the end to end connection, allowing the resources to be used in the most efficient way.
- Auto-Create Extensions
  IP Office can automatically create an extension entry for new IP telephones added onto the local area network. In cases where the local area network is not secure this facility can be disabled, but simplifies installation of IP telephone systems
- Fax Transport *(Avaya Proprietary)*
  Fax Transport allows fax calls to be routed over VoIP trunks between IP Office systems on an IP network using a proprietary IP Office transport protocol.
- Fax Transport T.38
  IP Office supports the standardized protocol "T.38" for transporting FAX calls between IP Office and SIP trunks or SIP endpoints. T.38 is only supported on IP Office 500 hardware. It requires the IP Office to be equipped with VCM 32 or VCM 64 modules.

  T.38 allows the reliable transmission of Fax messages over a IP channel independent of Codec used for the Voice communication.

Platforms Supported:	IP500, IP500 V2 only, must be fitted with VCM32 or VCM64 module.
Trunk Types Supported:	SIP
Extensions Supported:	SIP
T.38 Transport Layers Supported:	UDPTL (with optional redundancy error correction).
T.38 Versions Supported:	0-3
Call Types Supported:	Voice calls which transition to fax relay on detection of fax tones. Calls which are negotiated as fax only

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SIP Endpoint Support

Session Initiation Protocol (*SIP*)is an open signaling protocol for establishing any kind of real-time communication session. The communication session can involve voice, video, or instant messaging, and can take place on one of many devices that people use for communicating: laptop computer, PDA, cell phone, IM client, IP phone, and so on. SIP has been developed in the *Internet Engineering Task Force (IETF)* by common participation from various vendors, including Avaya.

Avaya IP Office supports SIP for telephony functions to enable the usage of standard based SIP endpoints for Voice and Fax communication. In pure SIP systems, IP Office expands the feature set beyond the SIP standard, offering a wealth of IP Office features also on SIP endpoints delivering a feature rich system that a pure-SIP server based on the SIP standard only can't deliver. With that, IP Office delivers the best of both worlds, supporting standard based IP telephones while delivering a wealth of features consistently between SIP, digital and Avaya IP endpoints.

IP Office SIP endpoint implementation is built on two major SIP components: SIP User agents, and SIP Server components.

SIP Components

SIP Endpoints (User Agents)

User agents (UAs) are applications in SIP endpoints (such as a SIP phone, cell phone, PDA, or workstation) that interface between the user and the SIP network.

SIP Servers

IP Office has implemented the required functionaliy of the SIP servers mentioned below not only to provide SIP endpoint support but also to allow full interoperability between SIP endpoints, other IP telephones based on H.323, Digital and Analog telephones as well as IP Office trunks (Analog, digital or SIP based)

SIP servers provide centralized information and enablement services in a SIP ecosystem. The core SIP servers and their functions are summarized here. IP Office provides the required the features of the following two servers for Voice and FAX communication.

- Registrar Server
  When SIP IP telephones come online, they need to make sure that others are aware that they're available to take and make calls. The Registrar authenticates and registers the IP phone (often directly related to a specific user) when it comes online, and then stores information on the telephones logical identities.
- Proxy Server
  A proxy server takes SIP requests, processes them, and passes them downstream while sending responses upstream to other SIP servers or devices. A proxy server may act as both a server and a client, and can modify a SIP request before passing it along. A proxy is involved only in the setup and teardown of a communication session. After user agents establish a session, communications occur directly between the parties.

Functionality of the following two SIP servers are generally available by IP Office using existing IP Office functionality. Therefore, while functionality is provided, e.g. allowing hotdesking (also for users using a SIP-endpoint) in a small community network, a consistent methodology between SIP and non SIP endpoints is used to deliver those features

- Location Service
  As users roam, the network needs to be continually aware of their locations. The location service is a database that keeps track of users and their locations. The location service gets its input from the registrar server and provides key information to the proxy and redirect servers. IP Office provides hotdesking support, delivering a similar functionality but working consistently between SIP and non SIP endpoints.
- Redirect Server
  If users are not in their home domains, sessions bound for them needs to be redirected to them.

  The redirect server maps a SIP request destined for a user to the device "closest" to the user. In IP Office, call forwarding and Follow me functionality is used to provide again consistent functionality between all type of endpoints.

Supported functionality for SIP endpoints in IP Office

SIP endpoints are supported on IP Office for Voice (Audio) and Fax (T.38) communication.

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This allows the usage of standard compliant IP telephones using the open SIP standard, giving customers a choice of endpoints of different manufacturers including special purpose devices like conference telephones, hotel telephones or terminal adapters.

In order to use a non-Avaya SIP endpoint with IP Office, a "Third party IP endpoint license" is needed. This license will continue to support endpoints based on the H.323 standard but will also be required for generic SIP endpoints on IP Office. Avaya IP Office SIP telephones use the IP End Point license.

SIP Endpoint support is fully integrated into IP Office core. No other components are needed. SIP endpoints will need VCM module capacity in IP Office like any other IP phone.

Next to SIP telephones, SIP terminal adapters are supported to connect analog telephones and fax machines. This offers a flexibility to support Fax machines and Audio/T.38

SIP extensions function like any other IP Office extension: This means they

- Can make and receive calls to any other extension, independent of type of extension
- Delivers end to end Media just like any other IP telephone on IP Office. For calls between two SIP extensions of a SIP extension and an Avaya IP telephone, the audio is transmitted end to end for basic telephone calls. (Conferences etc. However require a VCM resource). See chapter "VCM modules" for details
- Can use short codes and authorization codes like any other telephones
- Transmit In band call progress tones are delivered from IP Office
- A SIP telephones needs to register with IP Office like any other IP telephone, Authentication with Username and password is possible
- SIP extensions support "auto create" in IP Office to make installation fast and efficient. Successful registration of an endpoint will consume one third party license
- On one IP address, several extensions can register with IP Office, each consuming a license. This enables the connection of SIP terminal adapters with more than one analogue port, giving a different extension number to each of the ports.

Advanced features:
SIP endpoints support a number of extended features according to the "SIP service samples-draft", also referred to as "Sipping-19". This includes:

- Calling line identification
- Hold/Consultation Hold
- Attended/Unattended Transfer
- Message Waiting
- Do not disturb
- Conference Add

Some telephones support several call appearances making it easy to switch between calls. Please not that this does not include "bridged appearances" or " (outside)-line appearances)

A large number of additional features are supported on IP Office using Feature activation keys. These features include but not limited to:

- Call forward: Unconditional/Busy/no Answer
- Follow me
- Park/Unpark
- Music on Hold
- Meet me conferencing
- Conference join
- Ring back when free

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SIP endpoints also support Computer Telephony Integration “CTI” and therefore applications like One-X portal for Small business: the following features are supported with One-X portal and via the TAPI interface:

- Outgoing call *(without remote activation of speakerphone/headset)*
- Hang up
- Hold
- Attended/Unattended transfer
- Conference *(IP Office based)*
- Voicemail collect
- Set forwarding/DND *(IP Office based)*
- Park/Ride *(IP Office based)*

The features work similar like "CTI" features in combination with an analog telephone, e.g. an outgoing call will first ring the SIP phone and after connect the outgoing call will be placed. Avaya Phone Manager/ Phone Manager Pro and Soft console are currently not supported in combination with SIP-endpoints.

Session Initiation Protocol

. Rec. E.164 [2] - ITU-T Recommendation E.164: The international public telecommunication numbering plan
. RFC 2833 [7] - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals

. RFC 3261 [8] - SIP: Session Initiation Protocol
. RFC 3263 [10] - Session Initiation Protocol (SIP): Locating SIP Servers

. RFC 3264 [11] - An Offer/Answer Model with Session Description Protocol (SDP)
. RFC 3323 [14] - A Privacy Mechanism for the Session Initiation Protocol (SIP)

. RFC 3489 [18] - STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
. RFC 3824 [24] - Using E.164 numbers with the Session Initiation Protocol (SIP)

. RFC 1889 – RTP
. RFC 1890 - RTP Audio

. RFC 4566 – SDP
. RFC 3265 - Event Notification

. RFC 3515 - SIP Refer
. RFC 3842 - Message Waiting

. RFC 3310 – Authentification
. RFC 2976 – INFO

. RFC 3323 - Privacy for SIP (PAI) and draft-ietf-sip-privacy-04 (RPID)

For codecs supported, refer to chapter to VoIP Standards Supported.

While great care has been taken to be compliant with SIP standards, no guarantee can be given that all devices claiming support of SIP will work flawlessly. The SIP standard is constantly evolving with new features and methods introduced. Also, while being standard compliant, not all devices implement all options of the standard, making it hard to almost impossible to predict if a device will work- Avaya will only confirm functionality of devices that we have tested and will publish a list of devices that have been tested including – if required – implementation details and software version used on that device.

As of time of writing, the following devices have been tested successfully for Audio and/or Fax transmission.

SIP Telephones:	• Polycom Soundpoint IP 320, IP 601 • Grandstream GXP 2000, GXP 2020
SIP clients on mobile cell telephones:	• Nokia S60 v3 SIP client (e.g. Nokia E61i mobile cell phone)
SIP Analog Terminal Adapters	• Quick Edition A10 ATA • Patton single line M-ATA • Innovaphone IP22, IP24, IP28
SIP PC-based softphones:	• CounterPath eyebeam/xlite
Softphone	• IP Office Video Softphone (Using advanced SIP features): Does not require a “SIP-station” license but works as part of the user package

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Video enabled devices	• Avaya 1010 Video Avaya 1040 Video with 4-party MCU Polycom VVX 1500 Grandstream GXV3140

A list will be provided in the IP Office knowledge base and on http://support.avaya.com

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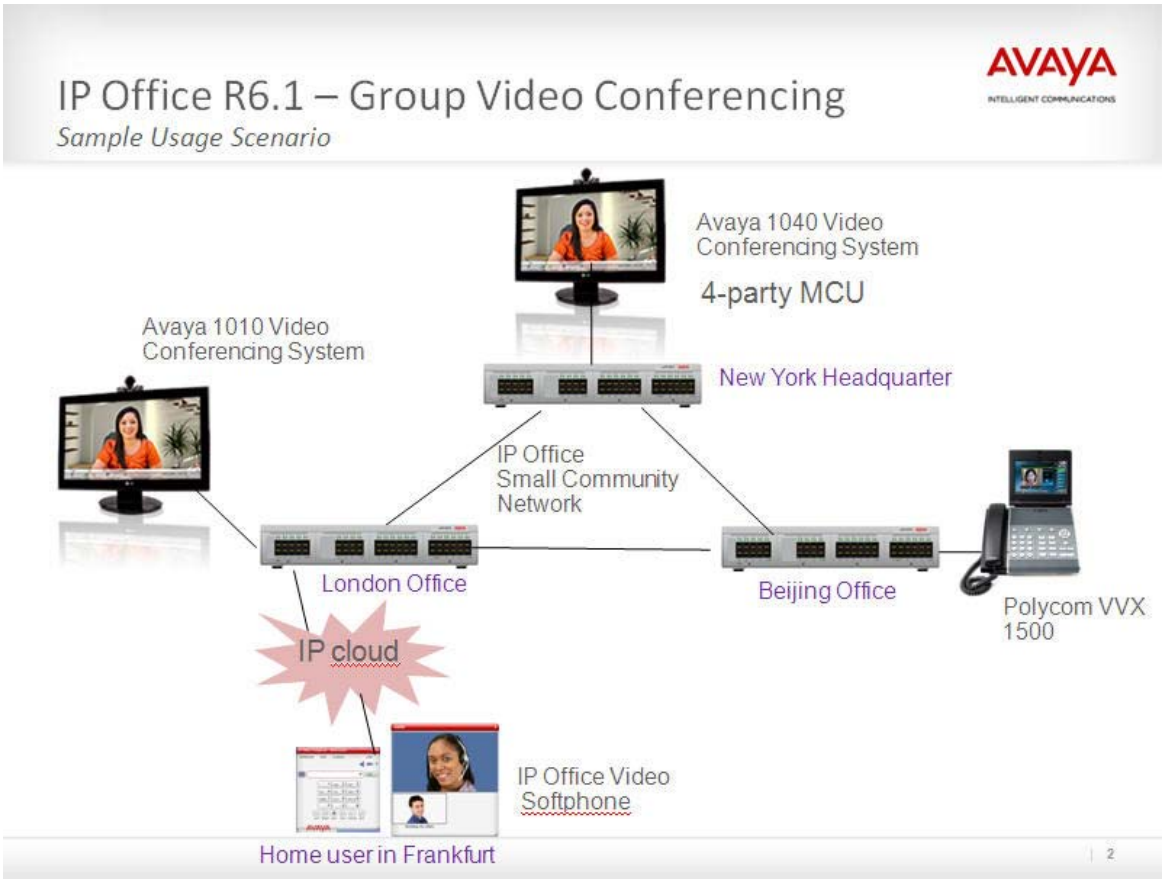
Supported functionality for SIP endpoints in IP Office

IP Office – since release 6.1 supports both Video and Audio communication for SIP devices.

It is therefore possible to connect a Video-endpoint to IP Office for rich multimedia communication:

Video is supported

- In an IP Office system locally
- In a IP Office small Community network
- To a Video ready SIP trunk, e.g. to Avaya Aura
- 

A few typical Call scenarios:

- End to end Video call in an IP Office system: E.g. a call from a Video Softphone to an Avaya 1010 system in a senior manager's office.
- Video in a Small Community Network: Calls e.g. between two Avaya 1010 telephones in the headquarter and a remote Office on the SCN
- Video Conference leveraging the Avaya 1040 Video Conferencing system. One user can use the 1040 while three other users are e.g. in remote locations on a SCN, home office etc.

Video details:

- All Video communication is end to end, meaning that IP Office does not natively manage or perform a Video conferencing
- Video endpoints can make calls to all telephones and trunk lines on IP Office (analog, digital IP) but will work as audio devices only when calling these telephones. They can also receive audio calls from these telephones/lines.
- Video needs a high bandwidth in the network, depending on codec/Video quality it can be up to 1 Mbit/sec. Through planning of the capacity it has to be assured that there is sufficient bandwidth available for both audio and video. A network assessment is mandatory and has to take the bandwidth requirements into account. Please refer to the details of the product(s) to be used for exact Video requirements.

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- All Video connections are “end to end”. IP Office does not negotiate e.g. Video Codes; this will be done by the Video endpoints (via IP Office) only. The Video-capability is therefore depending on the used devices. It was confirmed that IP Office works with both H.263 as well as H.264 Codecs
- Video calls have limited functionality. While with most devices hold and transfer works, this is of course depended on the tested device. IP Office feature access codes are available for Video telephones and e.g. call forwarding will function with Video devices. Forward to voice mail will work but will only record audio streams, no Video calls.
- IP Office supports a single Video call per session only. No dual Video channels for e.g. Video and sharing of applications are possible.
- A single endpoint can accept several video calls in parallel. That makes it possible to leverage “MCU” (Multi conference Unit) functionality in an endpoint like inside the Avaya 1040 system. It makes Video conferences between a device with MCU and several other endpoints possible.
- Typical bandwidth requirements for HD video:

  Video bandwidth needed per call (transmit and receive, Audio bandwidth not included) for e.g. the Avaya 1010/1040 Video Conferencing systems.

  - 1010: 1 Mbps for 720p/30fps
  - 1040:
    - 768 Kbps for 720p 30fps
    - 1.1 Mbps for 720p 60fps
    - 1.7 Mbps for 1080p 30fps

Power Options for IP Telephones

Avaya supports the IEEE 802.3af, standard for Power over Ethernet (PoE) on its range of IP telephones. With Power over Ethernet, both power and data are carried over one CAT 5 Ethernet cable. Deploying IP telephones utilizing Power over Ethernet eliminates the need for local power supplies, AC adapters and cables, and allows power to be provided from the wiring closet/switch room where it can be easily connected to a UPS system.

There are several power options, in addition to IEEE Power over Ethernet, available to customers to power their Avaya IP telephones.

- Avaya individual Power Supply for 4600 and 5600 telephones
  Individual power supplies that can be used to power each IP phone which provides a single 48 volt output. The power supply can operate globally within a wide range of Alternating Current (AC) input voltages: 90 - 264 Volts Alternating Current (VAC), 47-63 Hz. This power supply has a green indicator (LED) that shows the unit has power to the PHONE socket on pins 7&8 of the CAT5 cable. This item is available in two different versions, with and without an internal battery for uninterrupted power to the phone.
- Avaya individual Power Supply for 1600 Series telephones
  The 1600 Series telephones have a special power connecter to allow the connection of a cost efficient power supply. These power supplies come in several versions for different countries and voltages.
- Avaya Power over Ethernet adapter for the 1603 phone
  The 1603 phone does not support Power over Ethernet natively. A small adapter is needed. The adapter will fit completely in the stand of the phone, offering a cost efficient option of either using a power supply or a PoE adapter.

IP Telephone Power Consumption

Measured in Watts using an IEEE 802.3af power supply at 48V. This table shows the power consumption of the telephones currently shipping. Older versions of telephones might have a different power requirement.

		Power Consumption (Watts)		
Telephone	PoE Class	Conservation Mode	Typical	Max Power
4602SW, 5602SW	2	4.1	N/A	5.0
4610SW, 5610SW	2	4.0	N/A	6.0
4621SW, 5621SW	2	4.9	N/A	6.45
4625SW	3	7.8	N/A	9.42
1603/1603SW	2	4.32	3.75	5.10
1608	2	4.66	3.76	5.69
1616	2	3.17	2.83	3.65
Gig Adapter	–	3.16	N/A	5.81
9620L	1	2.1	2.2	2.7
9620C	2	4.6	4.9	5.3
9630G	2	4.6	4.9	6.1

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		Power Consumption (Watts)		
9640	2	4.2	4.6	5.5
9440G	2	4.5	4.9	5.6
9650	2	4.6	5.2	5.6
9650C	2	4.6	5.2	5.6
BM24	n.a	0.39	0.6	0.9
9608	1	1.93	2.08	2.55
9611	1	2.64	3.12	3.78
9621	2	3.18	3.49	4.72
9641	2	3.28	3.44	4.12
BM12	n.a	0.10	0.31	0.62

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Avaya Individual Power Supply

The power supply can operate globally within a wide range of Alternating Current (AC) input voltages: 90 - 264 Volts Alternating Current (VAC), 47-63 Hz. This power supply has a green indicator (LED) that shows the unit has power to the PHONE socket on pins 7&8 of the CAT5 cable.

This item is available in two different versions, with and without an internal battery for uninterrupted power to the phone.

1151 local power supply, without battery and with battery backup

For 1600 Series telephones, a separate line of cost efficient power supplies is available in different regional versions.

VoIP FAQ

Network Requirements

Quality of Service (QoS) is a measure of the performance of a network that reflects the availability of network service and the quality of network transmissions. The term itself refers to a number of networking technologies and techniques and does not necessarily restrict itself to any single protocol or standard.

There are a number of measures that can be taken on the LAN and WAN to make them 'good enough' to carry voice traffic. Some of these are the implementation of standards based QoS protocols while are simply a matter of network architecture and good network management practices.

The term 'good enough' is intentional. Every customer will have different expectations and different budgets to work to. Some will be willing to upgrade their networks to use the best possible equipment and practices. To others the additional expense may be viewed as unnecessary.

Examples of standards based Quality of Service protocols include DiffServ and 802.1p/q.

What are Voice Compression Modules (VCM's) for?

VCM's are required to support the following scenarios:

- Internal phone calls between an IP device and a non-IP device.
- Analog/digital telephones to IP trunks (SIP/H.323) including managed Frame Relay and managed IP VPN (provides echo cancellation).
- IP telephones to ISDN or PSTN trunks (convert IP to TDM and vice-versa).

IP device to IP device
Call progress tones (for example dial tone, secondary dial tone, etc) do not require voice compression channels with the following exceptions:

- Short code confirmation, ARS camp on and account code entry tones require a voice compression channel.
- Devices using G.723 require a voice compression channel for all tones except call waiting.

When a call is connected:

- If the IP devices use the same audio codec no voice compression channel is used.
- If the devices use differing audio codecs, a voice compression channel is required for each.

"Direct Media" is a VoIP concept that circumvents resources (TDM bus, Gateway) within the system and improves the voice quality. If two IP devices are connected on the same system, a direct LAN connection between them will be established once the call has been set up (as long as they use the same Codecs). Direct media is supported between Avaya IP telephones and SIP telephones.

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It is possible for an IP device to be physically located at one site while being registered at a different site. In this case, even for VoIP across the WAN the VCM would not be used, as long as the two IP devices involved in a phone call are registered on the same system.

Bandwidth Required For Each Voice Call?

The bandwidth used varies depending on the compression method chosen. IP Office supports standards listed below. These will occupy approximately 10K and 13K of bandwidth respectively.

Use the following chart to choose the most appropriate compression algorithm for your available bandwidth.

Audio Codec	RTP Voice Data Payload	Packets per Second	LAN (bps)	% Overhead LAN	WAN (bps)	% Overhead WAN	Algorithmic Delay (milli-seconds)
G.723.1 (6.3K)	24 Bytes	33.33	20,800	225%	9,867	54%	80
G.729a	20 Bytes	50	29,600	270%	13,200	65%	40
G.711 (64K)	160 Bytes	50	85,600	34%	69,200	8%	20

Typical Bandwidth Required For Video Calls

This is for Video transmit only. To the Video part, the audio channel needs to be added.

- Video bandwidth needed per call (transmit and receive, Audio bandwidth not included) as sample for the Avaya 1010-1040 for HD-Video:
  - 1010/1020: 1 Mbps for 720p/30fps
  - 1030/1040:
    - 768 Kbps for 720p/ 30fps
    - 1.1 Mbps for 720p/ 60fps
    - 1.7 Mbps for 1080p/ 30fps

Typical Bandwidth for standard Video quality, for example with IP office Video Softphone

Resolution	Pixel	Typical Net Video Bandwidth Requirement	
		H.263	H.264
Low	176 x 144 (QCIF)	64 kbps	45 kbps
Standard	320 x 200 (QVGA)	162 kbps	114 kbps
High	640 x 480 (VGA)	776 kbps	545 kps

Acceptable Delay?

End-to-end delay should be 150 milliseconds or below.

How Many Simultaneous Calls Can I Get Down My Link?

The following chart illustrates the theoretical maximum number of simultaneous voice calls that can be delivered over a WAN for a given link speed. This does not take into account any bandwidth that may be required for data traffic between sites or the physical limit of VoIP calls for the specific version of IP Office in use.

The number of simultaneous voice calls can be in excess of the capabilities of the individual platform, where the calls transit the switch as data traffic. In this situation compression resources are not used but obviously must be catered for in the overall bandwidth provision.

Compression	G.723.1 (6.3K)	G.729a (8K)	G.711 (64K)
Algorithmic Delay (seconds)	0.08	0.04	0.02
Number of Calls			
- 64Kbps Link	6	4	0
- 128Kbps Link	12	9	1

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- 256Kbps Link	25	19	3
- 512Kbps Link	51	38	7
- 1Mbps Link	103	77	14
- 2Mbps Link	207	155	29

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What is the Maximum Number of Simultaneous VoIP Calls

Each IP Office can be fitted with optional Voice Compression Modules (VCM) to support VoIP connections.

The IP500 V2 is capable of supporting two VCM 32/64 modules allowing up to 128 simultaneous calls. With combination cards, another 10 VCMs are provided per card. Up to two combination cards are supported with IP Office. of supporting two VCM 32/64 modules allowing up to 128 simultaneous calls.

Does the IP Office support Fax over IP ?

The IP Office supports a proprietary method for carrying Fax calls over IP between IP Office systems. It also supports T.38 on SIP trunks and SIP extensions.

Network Assessment

With IP Office, optimum network configurations can support VoIP with a perceived voice quality equivalent to that of the Public Switched Telephone Network (PSTN). However, not every network is able to take advantage of VoIP transmissions. It is important to distinguish between basic compliance with the minimal VoIP standards and validated support for QoS which is needed to run VoIP applications over a data network.

With the exception of standalone configurations where IP telephones connect directly connected to the ports on IP Office, Avaya now requires that all customers formally audit their networks for IP telephony readiness before attempting to install any VoIP application.

A network assessment should normally include:

- Physical inventory of all equipment inclusive of the current version of code, and configurations as needed.
- An accurate and complete network topology for all networked sites, inclusive of IP addressing and physical/logical connections.
- An evaluation of the network's topology to check that the design is both sound and reasonable.
- Measurement of packet loss, jitter and delay over the course of multiple days and measured on a per minute basis. A graphical representation of the data is the preferred output method.
- Examination of QoS/Class of Service (CoS) parameters in place in the network.
- Summary of findings and possible actions to correct problems.

The assessment should leave you confident that the implemented network will have the capacity for the foreseen data and voice traffic, and can support H.323, DHCP, TFTP, and jitter buffers in H.323 applications.

With this in mind, if you require support during or after an IP Office VoIP installation, a copy of your network assessment documentation will be requested by your support channel.

For more details about available tools, resources and services to enable you to audit your network for VoIP readiness, please contact your local Avaya representative.

IP Packet Flow Control

While a high-performance switch forwards data packets at full wire speed to and from its ports simultaneously, there may be times when a switch port may not be able to accept packets at the rate it is receiving them.

For example, the switch port may be receiving packets from multiple ports at the same time, or the switch port may be receiving packets from a port operating at a faster speed. For instance, the sending port might be operating at 100 Mbps, while the receiving port operates at 10 Mbps; or the sending port might operate at 1000 Mbps, while the receiving port operates at 100 or 10 Mbps. If data packets arrive for a port that is saturated with other packets, the packets may overflow the port’s buffer, resulting in dropped packets and lost data.

Flow control is a congestion-control mechanism that prevents data loss at congested ports. Flow control prevents packet loss by controlling the flow of data from the transmitting device to ensure that the receiving device can handle all of the incoming data.

IEEE 802.3 flow control is used on Avaya IP telephones operating in full-duplex mode. If the receiving device becomes congested, it sends a pause frame to the transmitting device. The pause frame instructs the transmitting device to stop sending packets for a specific period of time. The transmitting device waits the requested time before sending more data.

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VoIP Standards Supported

IP Office supports the following protocols and standards:

- H.323 V2 (1998), Packet-based multimedia communications systems.
- Q.931, ISDN user-network interface layer 3 specification for basic call control.
- H.225.0 (1998), Call signaling protocols and media stream packetization for packet-based multimedia communication systems.
- H.245 (1998), Control protocol for multimedia communication.
- Session Initiation Protocol.
- Audio CODECs:
  - G.711 A-law/U-law (64K).
  - G.723.1 MP-MLQ (6.3K).
  - G.729 Annex A, Annex B, Annex AB – CS-ACELP.
- Silence Suppression.
- Fax Relay (IP Office to IP Office Fax Transport over IP).
- T.38 Fax support (SIP trunks and SIP endpoints).
- Local End Echo Cancellation 25ms.
- Out of band DTMF.
- Jitter buffer, 5 frames of jitter buffer.
- Internet Standards/Specification (in addition to TCP/UDP/IP).
  - RFC 1889 – RTP/RTCP, Real Time and Real Time Control Protocol.
  - RFC 2507, 2508, 2509 – Header Compression.
  - RFC 2474 – DiffServ, Type of Service field configurable.
  - RFC 1990 - PPP Fragmentation.
  - RFC 1490 - Encapsulation for Frame Relay.
  - RFC 2686 - Multiclass Extensions to Multilink PPP.
  - RFC 3261 - Session Initiation Protocol (SIP).
  - RFC 3489 - STUN.

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Chapter 8.
Public and Private Voice Networks

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8. Public and Private Voice Networks

With Avaya IP Office you can be networked via T1, PRI and BRI ISDN, including VoIP on the company WAN. Networking maximizes the potential of your business office locations and remote workers - building the best possible foundation for your future growth. IP Office provides each location with a scalable (up to 384 extensions) telephony solution that supports voice networking, and offers:

- Uniform dialing plan, making it easy to call co-workers anywhere on the network and improve customer service
- Consistent user experience by sharing the same telephones and messaging interface as in headquarters
- A user-defined central directory that is automatically synchronized
- Automatic Route Selection to optimize routing and bandwidth management
- Centralized voicemail and/or the ability to network voicemail systems together

The benefits of networking:

- Leverage any existing investment in Avaya systems at other sites.
- Centralize services (e.g. operator, voicemail) as well as management and administration to reduce costs.
- Speed deployment of remote offices, respond quickly to market demands.
- Improve inter-site communication to simplify information exchange.

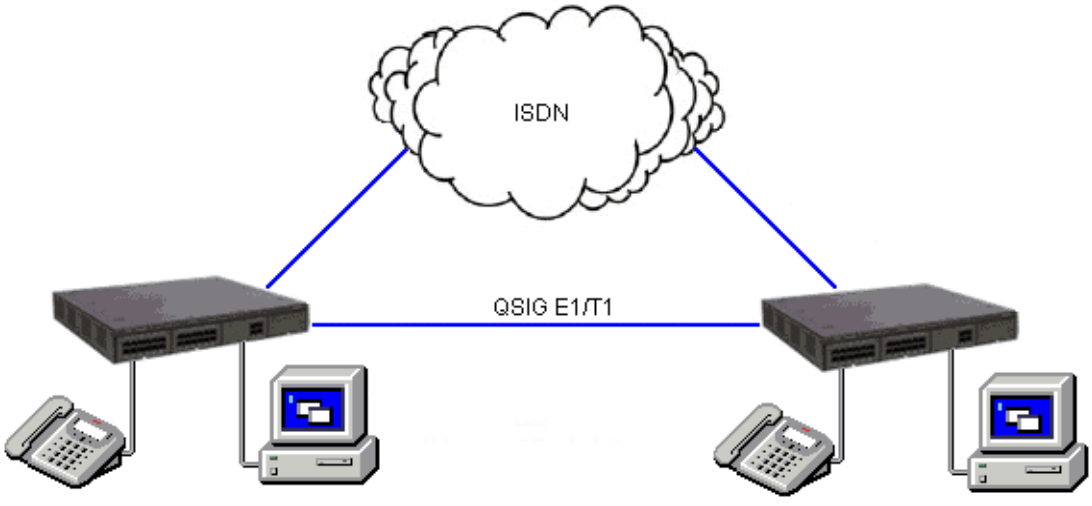
Private Circuit Switched Voice Networking

Private voice networks are built using structured leased line circuits (E1 or T1) or by establishing permanently connected 'B' channels over ISDN circuits between IP Office systems. Each channel within the E1 or T1 interface can provide a single voice or 64K/56K data call. Where leased line circuits are used within a private networking scenario, these E1 or T1 interfaces are typically configured to use QSIG signaling between sites.

QSIG provides a level of voice feature transparency between PBXs and is the favored signaling standard within multiple vendor and international voice networks. The IP Office E1 or T1 module terminates a QSIG connection with a 120 ohm RJ45 interface.

IP Office supports the following QSIG services across this network:

- Simple Telephony Call/Basic Call: ETS300 171/172.
- Circuit Switched Data Call/Basic Call: ETS300 171/172.
- Called/Calling Line ID Presentation: ETS300 173.
- Called/Calling Name Presentation: (SS-CNIP, SS-CONP, SS-CNIR) ETS300 237/238.
- Message Waiting: (SS-MWI) EN301 260/255.
- Transfer: (SS-CT) ETS 300 260/261.



Circuit Switched Voice Networking

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The IP Office platform supports a range of trunks and signaling modes for connection to the public switched telephone network (Central Office). Some of these lines are only available in certain territories; please check with your distributor for local availability. Primary rate trunks are available with either a single (24/30 channels) or dual trunk (48/60 channels).

ISDN Primary Rate (ETSI CTR4)

Provided by the IP400 PRI E1 and IP500 Universal PRI cards.

ISDN Primary Rate provides 30 x 64K PCM speech channels over an E1 circuit and one signaling channel. Signaling Conforms to the ETSI Q.931 standard with Cyclic Redundancy error Checking (CRC).

The following supplementary services are supported:

- Calling Line Identification Presentation (CLIP) provides the telephone number of the incoming call to the IP Office.
- Calling Line Identification Restriction (CLIR) prevents the telephone number of the IP Office being presented on an outbound call.
- Connected Line Identification Restriction (COLR) Inhibits the COLP service.
- Direct Dialing In (DDI) where the exchange provides the last x digits of the dialed number on an incoming call. This allows IP Office to route the call to different users or services.
- Sub-addressing Allows the transmission/reception of up to 20 digits, additional to any DDI/DID or CLIP information, for call routing and identification purposes.

ISDN Basic Rate (ETSI CTR3)

Provided by the IP400 Quad BRI and IP500 BRI cards.

ISDN Basic rate provides 2 x 64K PCM speech channels and one signaling channel using Q.931 signaling and CRC error checking. Both point to point and point to multipoint operation is supported. Multipoint lines allow multiple devices to share the same line; however point-to-point is the preferred mode.

Basic rate supports all the services that are supported on the primary rate version with the addition of

- Multiple Subscriber Number. This service is usually mutually exclusive with the DDI/DID service and provides up to 10 numbers for routing purposes, very similar to DDI/DID.

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Additional ISDN features

The following ISDN features are supported by IP Office 4.0 and later on both PRI and BRI trunks. Note that availability of these features is dependent on availability and support from the ISDN service provider, for which there may be charges.

- Malicious Call Identification – MCID
  (24xx, 46xx, 54xx, 56xx, 9500, 9600, T3, T3 IP, DECT telephones)
  Short codes and button programming features are available so that users can trigger this activity at the ISDN exchange when required. This feature is NOT available on standard ISDN DSS1 telephones.

- Advice of Charge – AOC
  (T3 digital and IP telephones only; Phone Manager)
  Advice of charge during a call (AOC-D) and at the end of a call (AOC-E) is supported for outgoing ISDN calls other than QSIG. The call cost is displayable on T3 telephones for call accounting purposes. The IP Office allows configuration of call cost currency and a call cost mark-up for each user.

- Call Completion to Busy Subscriber – CCBS
  (2400, 4600, 5400, 5600, T3, T3 IP, 9500, 9600, DECT telephones; Phone Manager)
  CCBS can be used where provided by the ISDN service provider. It allows a callback to be set on external ISDN calls that return busy. It can also be used by incoming ISDN calls to a busy user. This feature is NOT available on standard ISDN DSS1 telephones.

- Partial Rerouting – PR
  (2400, 4600, 5400, 5600, T3, T3 IP, 9500, 9600, DECT telephones; Phone Manager)
  When forwarding a call on an ISDN channel to an external number using another ISDN channel, partial rerouting informs the ISDN exchange to perform the forward, thus freeing the channels to the IP Office. This feature is NOT available on standard ISDN DSS1 telephones and it is NOT supported on QSIG.

- Explicit Call Transfer – ECT
  (The normal usage of this feature is by a third party application)
  ECT is supported on the S0 interface. A Call to an S0 Endpoint can be transferred to any other device such as an analog, digital or IP endpoint or to any trunk. The normal usage of this feature is by a third party application connected via one or more S0 interfaces to IP Office. One example is the VoiceDirector, an automatic call assistant.

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North American T1

Provided by the IP400 PRI T1 and IP500 Universal PRI cards.

T1 Primary Rate provides up to 24 64K channels over a 1.54M circuit. Each channel of the T1 trunk can be independently configured (channelized) to support the following signaling emulations with handshake types of immediate, delay or wink.

- Loop-Start
- Ground-Start
- E&M Tie Line
- E&M DID
- E&M Switched 56K
- DID - Channels configured for DID/DDI support incoming calls only. The carrier or Central Office will provide the last x digits that were dialed to be used for call routing.
- Wink-Start

IP Office T1 trunks support both DNIS and ANI services, where available from the central office.

- Dialed Number Identification String (DNIS) Provides a string of digits to the IP Office depending on the number dialed by the incoming caller. This string can then be used to route callers to individual extensions, groups or services.
- Automatic Number Identification (ANI) Provides IP Office with a number identifying who the caller is. This may then be used for routing or computer telephony applications.

T1 trunk cards incorporate an integral CSU/DSU, eliminating the need for an external unit. The CSU function allows the trunk to be put in loop-back mode for testing purposes. This can be set manually, using the monitor application, or automatically from a Central Office sending a Line Loop Back (LLB) pattern. The DSU function allows the T1 trunk to be shared between data and voice services.

North American Primary Rate Interface

Provided by the IP400 PRI T1 and IP500 Universal PRI cards.

IP Office supports Primary Rate ISDN trunks on 5ESS or DMS100 central office switches provided by AT&T, Sprint, WorldCom and other Local Telcos. Channels can be pre-configured for the supported services or negotiated on a call-by-call basis.

Special Services can be configured to route calls to local operators or pre-subscribed carriers for both national and international calls (SSS). Alternate carriers can also be selected through the configuration of IP Offices Transit Network Selection (TNS) tables.

IP Office also supports the Calling Name and Number service over Primary Rate Trunks (NI2).

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Analog Trunks

- Loop Start
  Loop start trunks are available on the IP Office Analog Trunk cards installed within the IP Office control unit, or on the Analog Trunk 16-port expansion modules (ATM16). The first two trunks on the ATM16 are automatically switched to power fail sockets in the event of power being interrupted. They conform to the TIA/EIA-646-B standard. The loop start trunks also support incoming caller line identification (ICLID) conforming to GR-188-CORE and GR-31-CORE standards. IP Office can use this information to route calls or provide it to computer applications to display additional information about the caller.

- Ground Start
  Ground Start trunks are only available on the ATM16, configured through IP Office Manager. The first two trunks on the module are automatically switched to power fail socket in the event of power being interrupted. They conform to ANSI T1.401 and TIA/EIA-646-B standards. Not available in all territories.

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E1R2 Channel Associated Signaling

Provided by the IP400 PRI E1R2 and IP500 Universal PRI cards.

The IP400 PRI E1R2 cards are available in two versions supporting either RJ45 or coax network connections. Each card provides channels that can be configured for MFC, Pulse or DTMF dialing dependent on the requirements of the network.

Session Initiation Protocol (SIP)

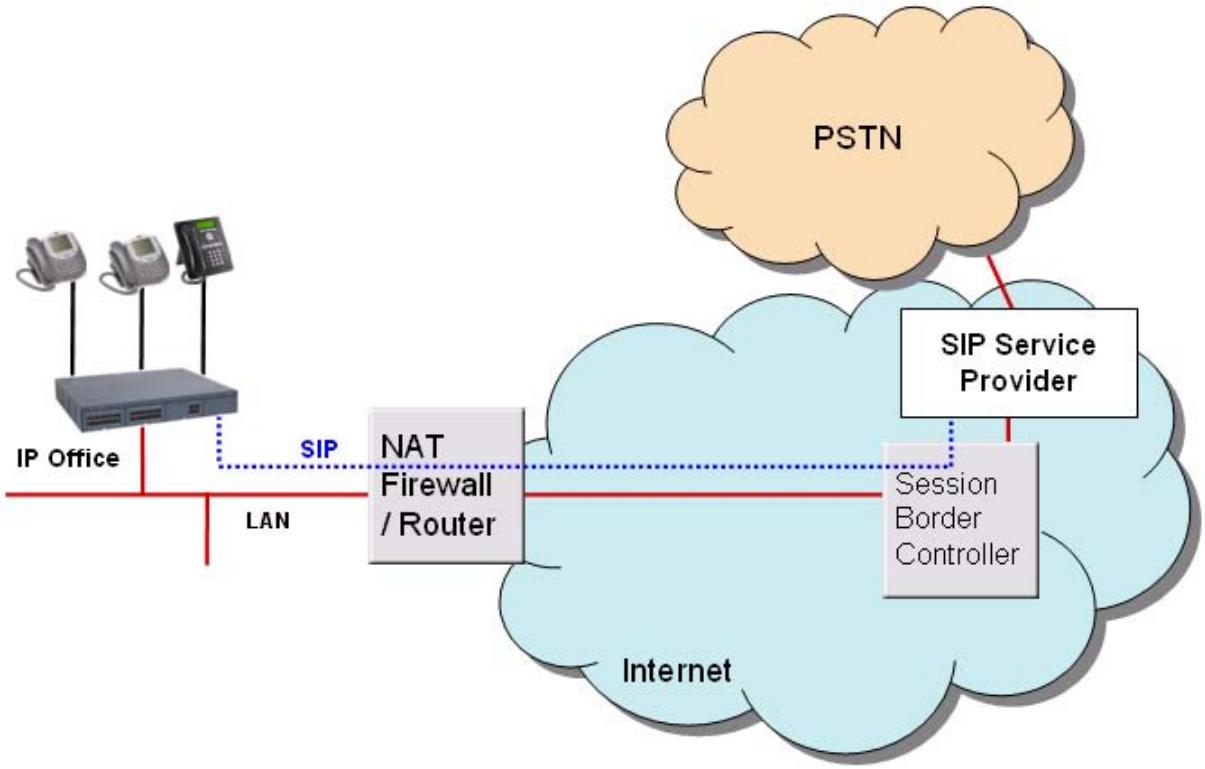
IP Office supports SIP trunking. SIP trunks allow IP Office users to take advantage of new telephony services being offered by 'Internet Telephony Service Providers (ITSPs)'. In many cases, these telephony services can offer substantial savings in comparison to traditional exchange lines. The IP Office solution allows all users, regardless of their phone type, to make and receive SIP calls. SIP trunks are handled like any other line on IP Office, affording all the call routing and toll control needed to manage inbound and outbound calls.

SIP trunks on IP Office require the provisioning of voice compression channels through the installation of VCM modules within the control unit. RTP Relay is also supported to allow the IP stream through SIP after call setup. A license for the maximum required number of simultaneous SIP calls is also needed. Up to 128 simultaneous calls are supported.

IP Office also supports the T38 protocol which allows Fax over IP.

There are several possible network topologies for SIP trunk systems, as shown in the following diagrams.

Option 1: Service provider with a Session Border Controller (SBC), which solves NAT traversal issues – this is the most reliable and preferred method.



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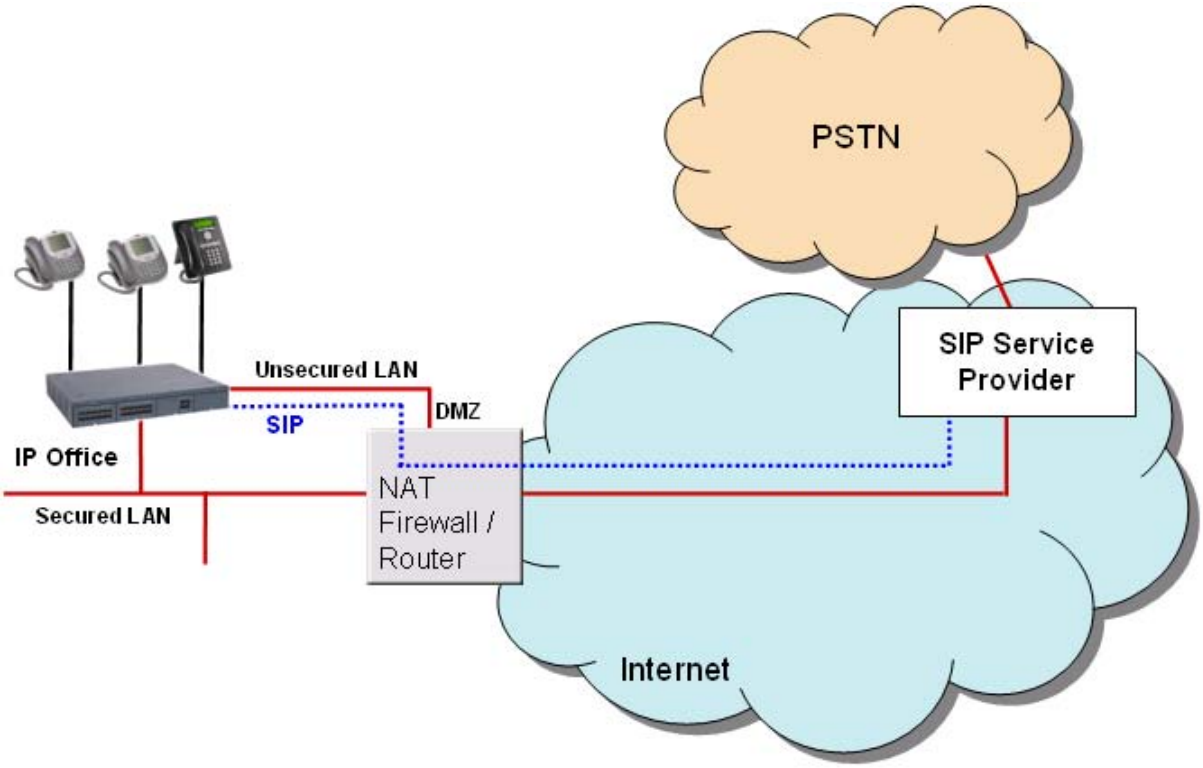
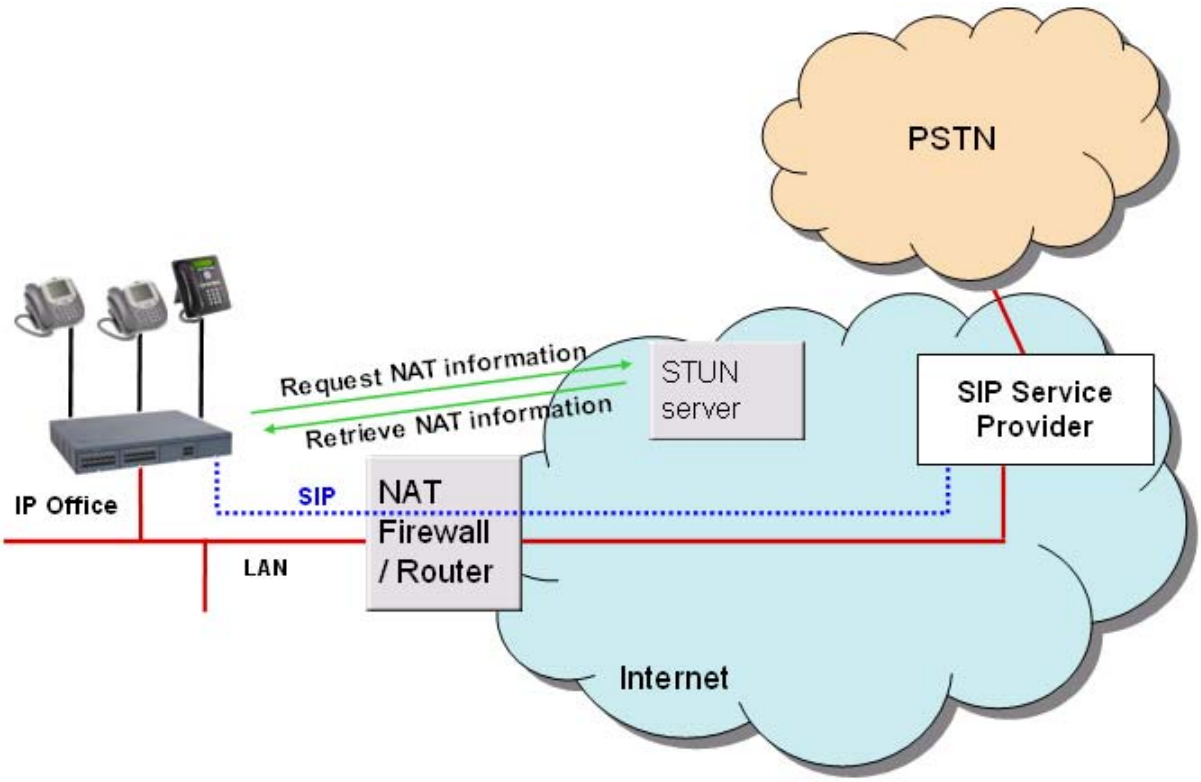
Option 2: Direct connection from the IP Office's second Ethernet port to the internet via a DMZ (demilitarized zone) port on the router. To make this configuration secure, the IP Office firewall is set to drop all packets except SIP.



Option 3: Connection to the ITSP over NAT using 3rd party STUN (Simple Traversal of UDP through NAT) servers in the network to discover the NAT mechanism being used.



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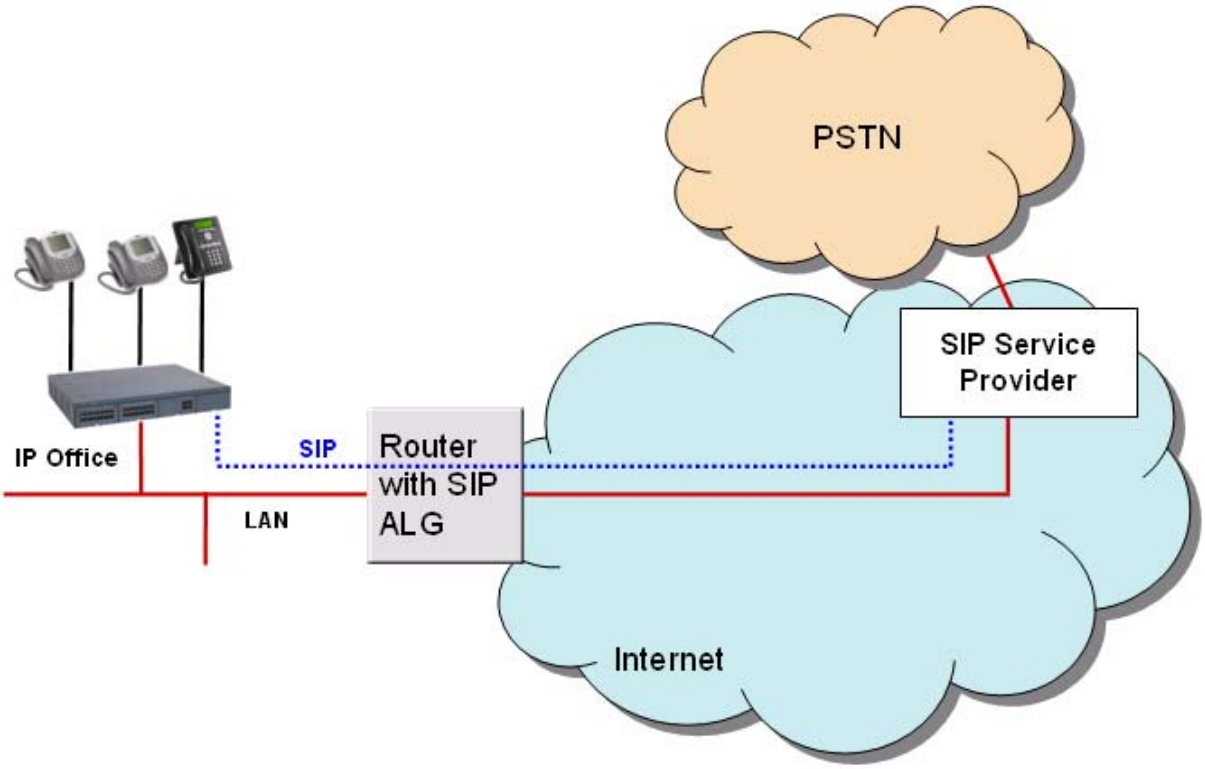
Option 4: Connection to the ITSP through a router equipped with an Application Level Gateway (ALG) which transparently resolves all NAT issues.



For details on SIP ITSPs which have been tested by Avaya, please see the Technical Bulletin for the IP Office 4.0 release and/or IP Office Knowledge Base at http://www.avaya.com/ipoffice/knowledgebase.

Reference ITSPs supported in IPO Office Releases 5.0 and higher at https://devconnect.avaya.com/

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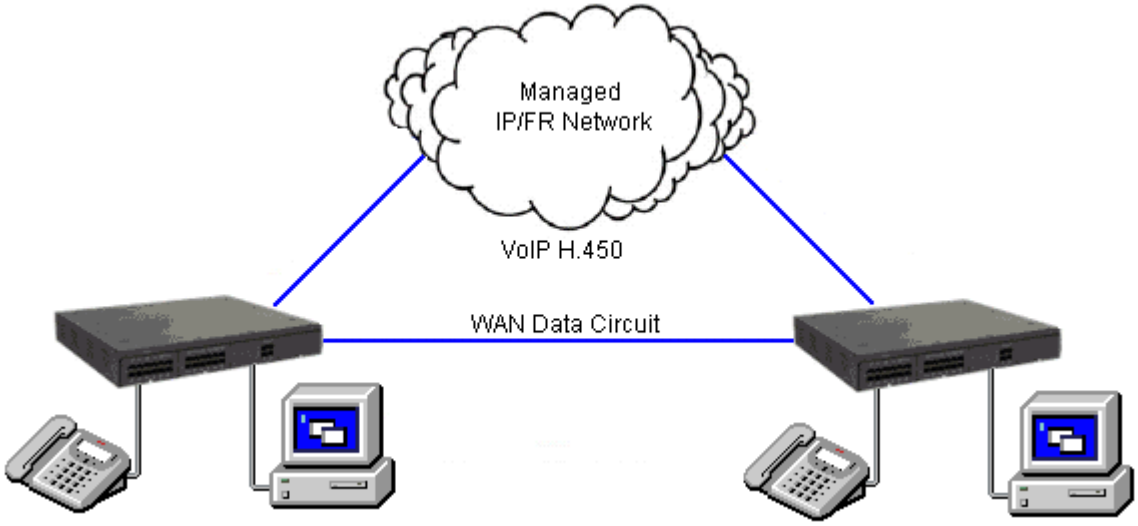
Packet Based Voice Networking

This section describes the options available for businesses that are able to utilize data networks to support voice solutions such as Voice over IP (VoIP). IP Office offers networked voice and data communications, providing:

- Built-in IP router
- One link for voice and data networking
- Common access to the Internet; share files and send e-mails to other sites
- Support for RIP-2 protocol for dynamic data routing; IPSec VPN, firewall and NAT (Network Address Translation) and for security; Centralized management and proactive fault management via SNMP.

IP Packet based voice networking between IP Office sites can be achieved through:

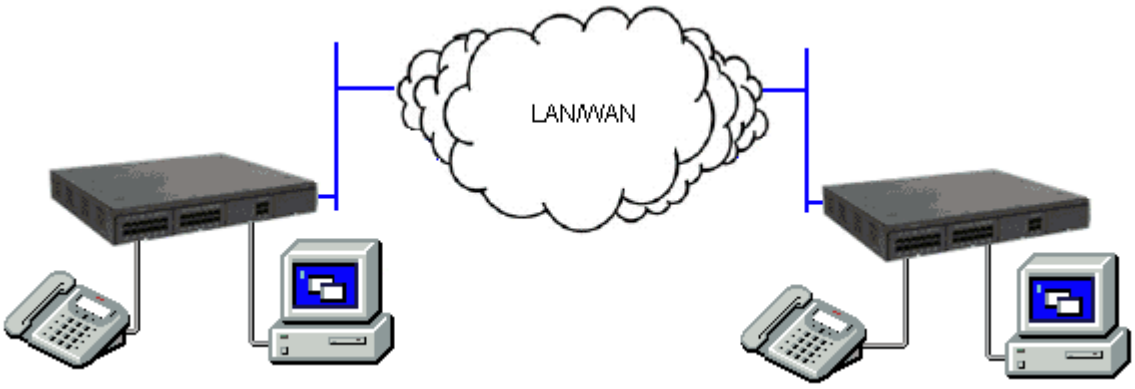
- VoIP across the campus LAN
- or VoIP across the public network.



VoIP networking across IP network or WAN

VoIP across the LAN

In a factory or campus environment, voice calls can sent over 10/100 Mbps LAN connections on systems equipped with optional Voice Compression Modules (VCM). In order to avoid bandwidth contention issues, VoIP across the LAN will require some form of bandwidth management through Diffserve.



VoIP networking across the LAN

VoIP across the Public Network

IP Office is capable of realizing the benefits of Q.931 and H.450 supplementary service support across a public connection where an appropriate QoS connection can be established.

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Supplementary Services within IP Networks

Supplementary services within an IP environment are provided via Q.931 and H.323. IP Office provides the same rich services as enjoyed within a traditional network environment. Our standards based approach allows interoperability within mixed vendor networks.

The basic supplementary service features supported by H.323 on IP Office to IP Office IP trunk links are listed below.

- Basic call set up (voice).
- Call Hold (local).
- Call Transfer (local).
- Called/Calling Name.
- Called/Calling Number.

Additional features can be added through the use of IP Office Small Community Networking (see the following section).

On IP trunks to non-IP Office systems the Supplementary Service will depend on those also supported by the non-IP Office system.

Multi-Site Networking (Small Community Networking)

When connecting IP Offices together over IP or Packet based networks, Small Community Networking enhances feature transparency. These networks can support up to a maximum of 1000 users across 32 sites. The following additional features are available.

- Busy Lamp Field
- Camp-on
- Call Back When Free
- Paging
- Call Pick-up
- Centralized Personal Directory for 1400, 1600, 9600 and T3 Telephones as well as Avaya one-X™ Portal for IP Office
- Centralized System Directory for 1400, 1600, 9600 and T3 Telephones as well as Avaya one-X™ Portal for IP Office
- Centralized Call Log for 1400, 1600, 9600 and T3 Telephones as well as Avaya one-X™ Portal for IP Office
- Centralized Voice Mail (Preferred Edition)
  Support for mailboxes, call recording, dial by name and auto attendants. Remote queuing on remote systems is also supported.
- Distributed/Backup Voice Messaging
- Internal Directory
- Absence Text Message
- Anti-Tromboning.
- Distributed Hunt Groups
  Hunt groups can include users located on other IP Office systems within the network. Note that Distributed Hunt Groups are not supported for use with Customer Call Reporter.
- Remote Hot Desking
  Users can hot desk between IP Office systems within the network. The system on which the user configured is termed their 'home' IP Office; all other systems are 'remote' IP Offices. Note that Remote Hot Desking is not supported for use with CCC and CCR
- Breakout Dialing
  This feature allows the user to select an IP Office system in the network from a displayed list and then dial a subsequent number as if dialing locally on the chosen system. This feature is triggered either by a programmable button or short code.

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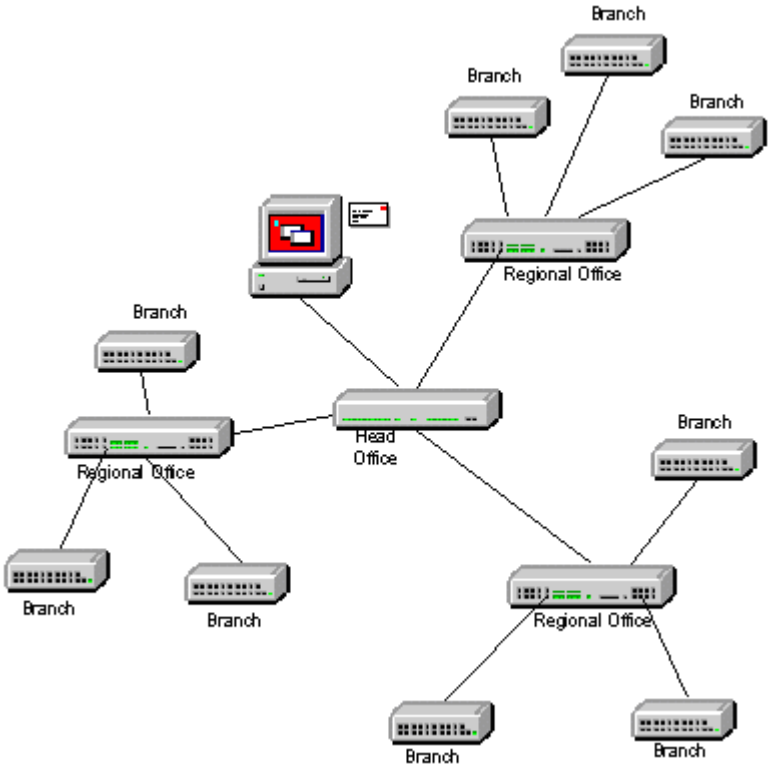
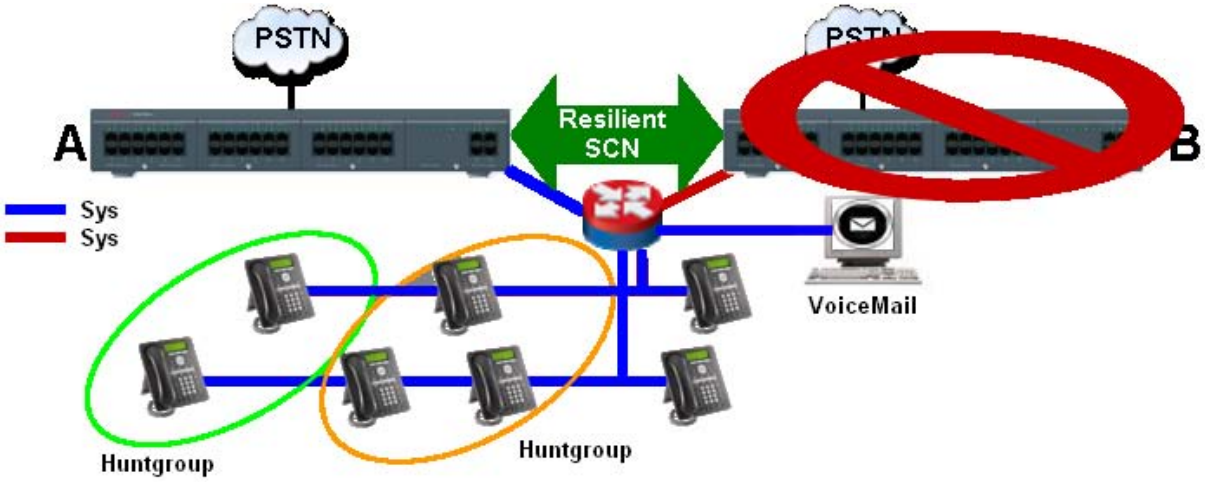


- Resilient Multi-Site Networking(SCN)(see the diagram below)
  In an SCN configuration of System "A" and System "B" where the centralized voicemail is connected to System B, and a number of IP telephones are connected to either System A or System B. If System "B" fails then:
  - System A will automatically take over from System B and support Avaya IP telephones, hunt groups, and DHCP if required.
  - VoiceMail Pro will reregister to System A.
  - For users in an Small Community Network (SCN), when they hot desk to another IP Office system, they retain their licensed profile setting as configured on their home system.
  - All System B users' Personal contacts/Call Logs will continue to be available (1600 Telephones).



For Multi-Site Networks (SCN), VCM modules are required in all systems being connected. The IP lines may be configured in a start or a meshed configuration. One of the advantages of a meshed configuration is that it removes the risk of a single point of failure within the network. Also the names and numbers (groups, line, services, etc) on the separate IP Office systems should be unique to reduce potential maintenance confusion.

Each IP Office system broadcasts UDP messages on Port 50795. These broadcasts typically recur every 30 seconds but BLF updates are potentially more frequent. There are no updates if there is no activity and the overall level of traffic is very low – typically less than 1 kbps per system.

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From IP Office Release 2.1(35) and higher, Multi-Site Networking is supported between IP Office systems with differing software levels but network features will be based on the lowest level of software within the network. This option is intended to allow the phased upgrading of sites within a Multi-Site network and it is still recommended that all systems within a network are upgraded to the same level where possible.

If larger networks are required QSIG can be used to link multiple Small Community Networks together. Functionality between the communities is governed by the QSIG feature set.

IP500 Voice Networking License
On IP500 and IP500 V2 systems, Multi-Site Networking (SCN) requires one or more additional licenses.

QSIG, H.323 and Multi-Site (SCN) capabilities are not enabled by default in the IP500 and IP500 V2. An additional license is required to enable this functionality with 4 simultaneous networking channels (no channel limit for QSIG). Additional channels can then be licensed in increments of 4. A Voice Networking license is still required to enable TDM QSIG, even though there is no limit to the number of TDM QSIG calls that can be made or received once licensed.

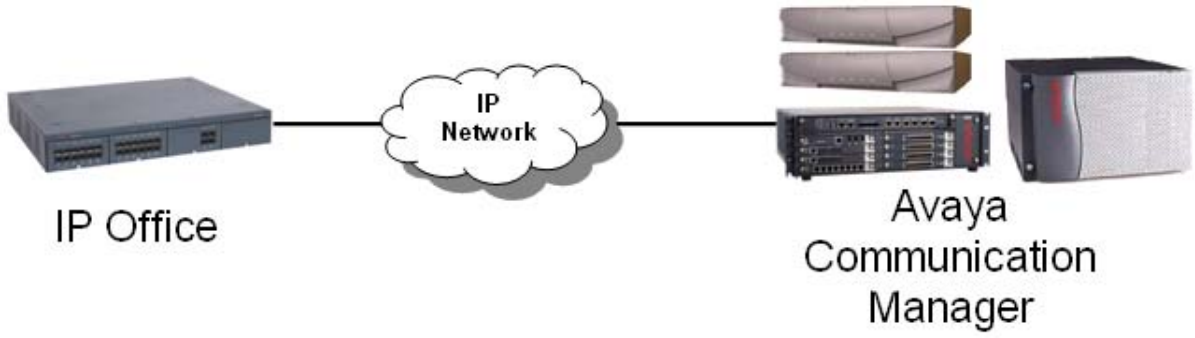
Internetworking with Other Avaya Products

IP Office will support the most appropriate way for communication with any other existing PBX's in a customer network, whether TDM or IP-based. With Avaya DEFINITY, MultiVantage, Avaya Integral Enterprise or Avaya AURA® Communication Manager (ACM), the protocols used will be QSIG or H.323 over T1, E1 or IP links

VoIP networking using H.323

IP Office (since release 1.1 in US and release 1.2 in EMEA) has been successfully tested to be interoperable over IP trunks with Avaya Communication Manager (ACM). The protocol supported is H.323 using industry-standard compression codecs (types G.711A, G.711MU, G.729A and G.723.1-6K3). The features currently supported are:

- Desk to desk dialing (basic voice call)
- Calling/Connected Party ID number
- Calling/Connected Name Presentation
- Call Hold
- Call Transfer



These features allow for simple networking needs between IP Office remote branches to a DEFINITY/ACM at the main site.

QSIG networking using T1/E1 links (TDM)

Alternatively QSIG may be favored as the chosen signaling standard within multiple vendor environments and provides the following supplementary services which are also available between IP Office and DEFINITY / MultiVantage/ I55 /ACM (equipped with the relevant RFA licenses):

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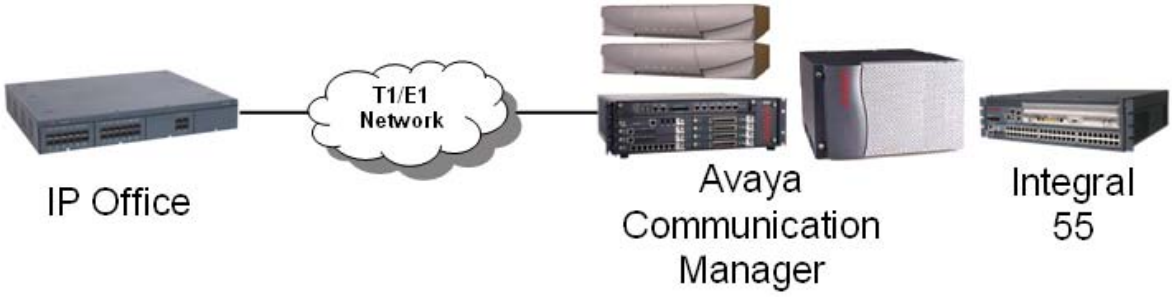

- Simple Telephony Call/Basic call (ETS 300 171/172)
- Circuit Switched Data Call/Basic call (ETS 300 171/172)
- Calling/Connected Line Identity Presentation (ETS 300 173)
- Calling/Connected Name Presentation (ETS 300 237/238)
- Message Waiting Indication (ETS 301 260/255)

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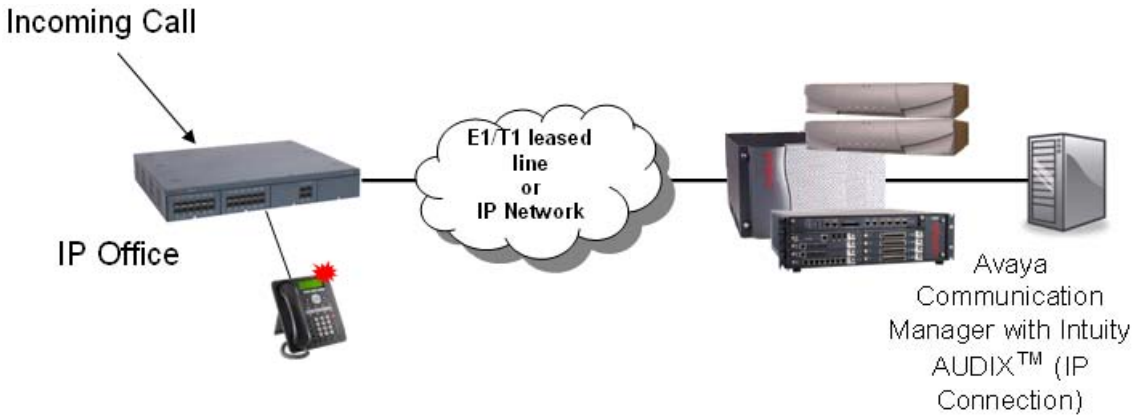
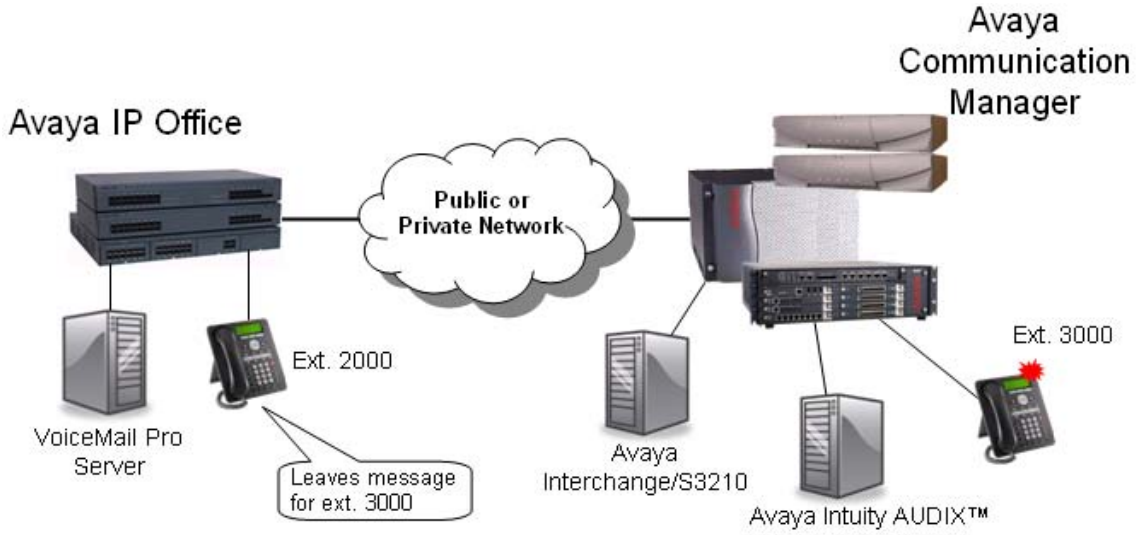
Messaging Networking

There are 2 options available today to provide messaging interoperability between IP Office and Definity / MultiVantage / ACM. The first option provides Centralized Voicemail while the second allows Avaya voicemail systems to be networked. The requirements, functionality and restrictions are summarized below:

- IP Office to DEFINITY / MultiVantage / ACM connected to Intuity AUDIX™ over a QSIG link (E1/T1 or IP)



- No local Voicemail required on remote branch IP Office but AUDIX license required on every IP Office
- Requires Intuity Audix 4.4+ connected via C-LAN to DEFINITY 9.5+ (see IP Office Offer Announcement dated August 2003 for more information on compatibility)
- Maximum of 19 IP Offices can be supported on 1 INTUITY AUDIX™ server (20 total with DEFINITY/ACM occupying one slot)
- Requires QSIG and Private Networking licenses on DEFINITY / MultiVantage / ACM
- User mailbox with Message Waiting Light support
- Forward voicemails between users
- No auto attendant (enhancement currently being investigated)
- No call recording
- No queuing at remote sites
- No Fax over IP to AUDIX™
- No Multi-Site Networking (SCN) support when AUDIX™ is configured on IP Office.
- Avaya IP Office VoiceMail Pro networked to Avaya Modular Messaging / Octel / Intuity AUDIX™ via Interchange / S3210



- Requires local VoiceMail Pro on every branch IP Office licensed with VoiceMail Pro Networked Messaging RFA
- Requires Avaya Interchange/S3210 on Modular Messaging, Octel or Intuity Audix
- Provides 2,000 remote mailboxes per VoiceMail Pro server i.e. per branch office (to be extended to 10,000 remote mailboxes by next VoiceMail Pro maintenance release)

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- User mailbox with Message Waiting Light support
- Forward voicemails between known remote users
- Fully-featured VoiceMail Pro at every branch office
- VoiceMail Pro Networked Messaging will only accept an incoming voicemail message for a local mailbox. It will NOT forward it to a remote Voicemail server. If required, this facility is available through Avaya Interchange.
- VoiceMail Pro Networked Messaging is limited to Avaya Messaging Servers (not third-party messaging platforms).

Common Networking Features

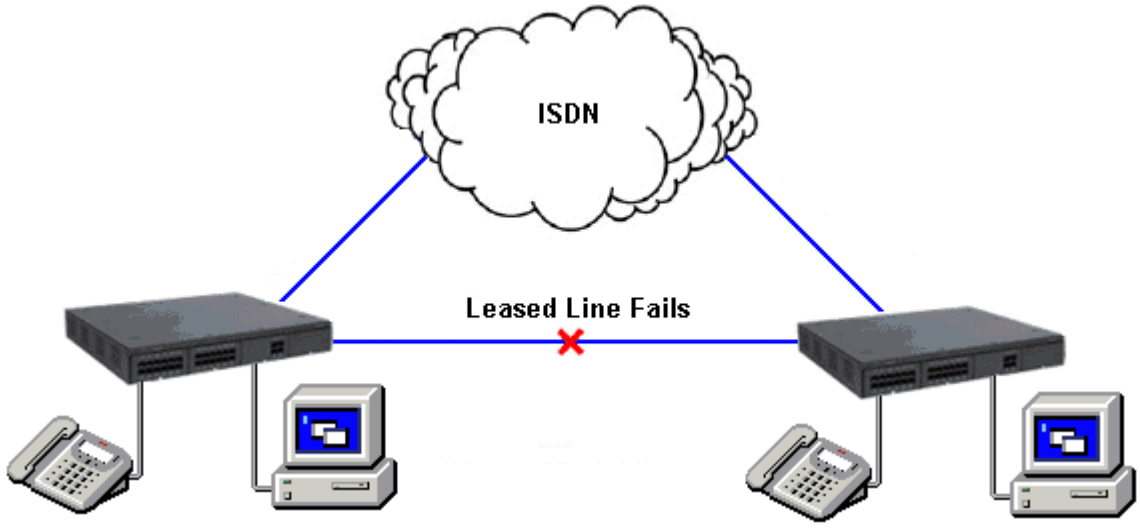
Alternate Route Selection

Alternate Route Selection (ARS) allows calls to be routed via the optimum carrier. Time profiles can also be used to allow customers to take advantage of cheaper rates or better quality at specific times of day.

If a primary trunk is unavailable or congested, then ARS provides automatic fallback to an available trunk (e.g., analog trunk fallback if a T1 or SIP trunk fails, or use PSTN for SCN fallback).

Multiple carriers are supported. For example, local calls are to go through one carrier between specific hours and international calls through an alternative carrier. Carrier selection using 2-stage call set up via in-band DTMF is possible. It is possible to assign specific routes on a per user basis, e.g. only allow expensive routes to be used by critical staff.

Note: Existing Least Cost Routing (LCR) configurations are automatically converted to ARS when upgrading to 4.1 or higher.



Network Numbering Schemes

IP Office uses fully flexible network numbering options. Dialed digits can be manipulated to add or remove digits, access codes etc. in order to fit into any numbering scheme. Two types of numbering schemes are commonly deployed - 'Linked Numbering' and 'Node Numbering' schemes. In linked numbering schemes each site within the network has a unique range of extension numbers and users simply dial the extension number of the called party. Often, linked numbering schemes are used in very small networks (< 5 sites) with less than 500 extensions. With node numbering schemes each site is given a node ID and this is prefixed by the user when dialing extensions at other sites. In this way extension numbers can be replicated across sites while still appearing unique across the network. Node numbering schemes are common in larger networks. Linked numbering schemes and node numbering schemes are sometimes both used within the same network with node numbering used at the large offices and linked numbering employed at clusters of satellite offices.

The following figures depict these two types of numbering schemes.

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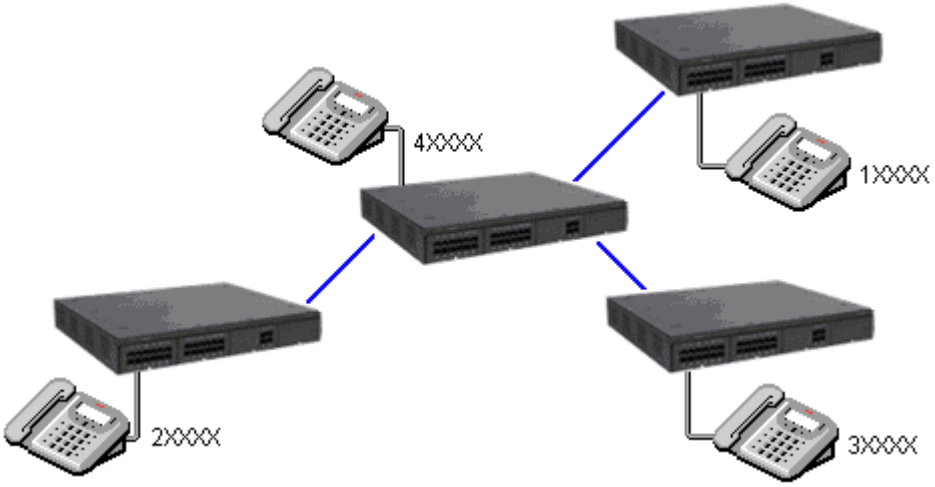
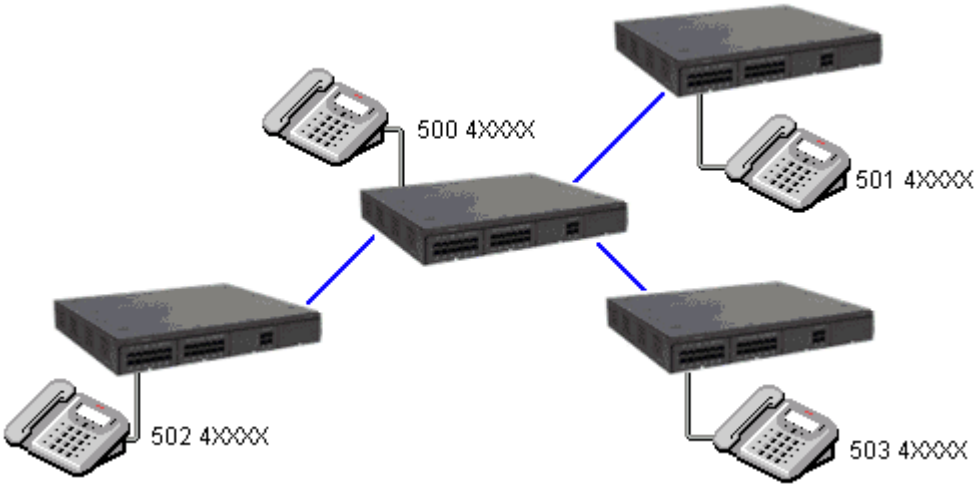


Linked Numbering Scheme



Node Numbering Scheme

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Chapter 9.
Data Networking Services

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9. Data Networking Services

LAN/WAN Services

The IP500 V2 supports a firewalled 2 port Layer 3 Ethernet Switch.

When computers on the LAN communicate they do not care where the destination is, they just send messages with the address of the destination. These messages are likely to be received at all other computers on the same network but only one – the target destination – will act on the message. Where the destination is on another network, the router is needed to be the "gateway" to the rest of the world and find the optimum route to send the message on to the destination. The router alleviates the need to establish and hold a call for the duration of a communication session (when messages or IP packets are being sent between source and destination) by automatically establishing a connection only when data is to be passed. Routers may be connected together using WAN (Wide Area Network) links that could be point-to-point leased lines, managed IP networks, Frame Relay networks or exchange lines (Central Office). The IP Office system supports all of these types of network connections.

IP Office has an integral router with support for bandwidth on demand that allows the negotiation of extra bandwidth dynamically over time. Where connection is over ISDN, IP Office initiates extra data connections between sites only when there is data to be sent or sufficient data to warrant additional channels. It then drops the extra channels when they are no longer needed. The calls are made automatically, without the users being aware of when calls begin or end. The rules for making calls, how long to keep calls up etc, are configurable within IP Office.

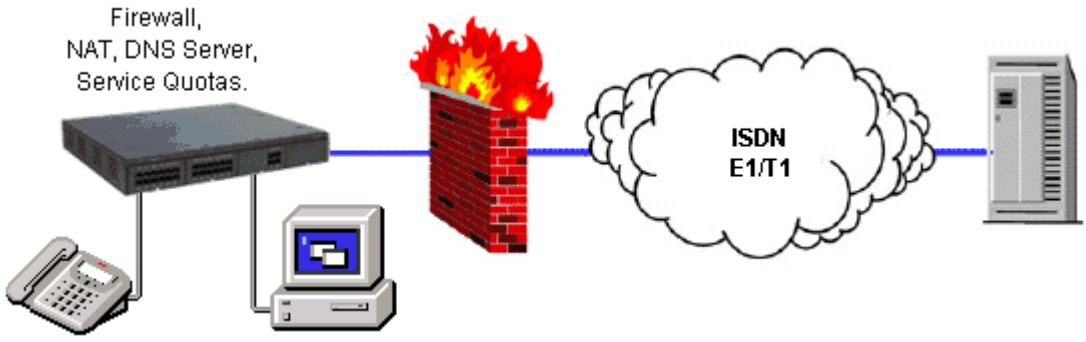
It is possible to have several different routing destinations or paths active at any time linking the office to other offices and the Internet simultaneously.

Internet Access

While the telephone is still the number one business communication tool, Internet access is becoming increasingly important for business-to-business communications. The ability to send and receive email, is now considered mandatory when dealing with many suppliers and customers, while access to the Internet for e-commerce applications and information has become vital.

IP Office systems provide shared, secure, high-speed access to the Internet via exchange lines (Central Office), digital leased line or IP VPN services.

Internet security concerns are addressed through the provision of an integrated firewall so removing the need for a standalone firewall. The firewall can be configured to cater for a variety of situations and will allow customers to control who can access external resources and when. The firewall isolates your private networks from the Internet, thereby ensuring that your network remains beyond the reach of hackers, while configurable service quotas can be set against a remote access service to ensure authorized users can gain access. Service Quotas place a time limit on outgoing calls to a particular IP Service so limiting costs. Each service can be configured with an alternative fall back, for example, you may wish to connect to your ISP during working hours and at other times take advantage of varying call charges from an alternative ISP. You could, therefore, set up one service to connect during peak times and another to act as fallback during the cheaper period.



Internet Access

Remote Access Features

IP Office's integral firewall, service quotas and timebands all apply to remote access calls. Remote access security can be supplemented by CHAP (encrypted passwords) to verify the end users, or PAP which does not support encryption. Timebands can control the hours within which the remote access service is available.

A "trusted location" can be set for dial in. These are locations that the System will allow either data access, e.g. a user dialing in from home, or access to voicemail without a voicemail code for a user collecting their voicemail messages from a mobile. The trusted location is also the location the Voicemail Server will call to inform the user of a new message.

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Conversely a "specified location" can be set which restricts remote access from only that location, this specified location can also be a designated dial back number thereby minimizing the threat of unauthorized remote access.

IP Office systems can also incorporate remote access dial back services so that if a user always remotely accesses the office from a single location e.g. their home, then after login verification the system will disconnect their call and dial them back. In addition to the added level of security dial back provides it can also be an excellent method of consolidating remote access charges onto the central office telephone bill.

In addition to remote access from Telephone Adaptors, all ATM4 trunk cards (including the IP500 V2 Combination Card ATM) support switching of the first analog trunk to an integral V.32 modem for remote access.

LAN to LAN Routing

All businesses now have a need for data routing whether it's a requirement to share resources such as email servers, file servers and internet gateways, or seamlessly transport data between sites or network to and from their customers and suppliers. This is why each IP Office platform offers IP routing as standard.

Embedding a router within IP Office removes the costs, complexity and additional points of failure of external WAN multiplexers by allowing data and voice traffic to converge and share the network resources of IP Office. These network resources can range from dial up ISDN connections, point-to-point leased circuits, managed IP networks or Frame Relay as IP Office supports all these types of network connections.

Data Networking Features

Integral 10/100 Mbit Layer 3 Ethernet Switch

Layer 3 switching is particularly useful in situations where it is desirable to have a ‘trusted’ and ‘unsecured’ network, where the ‘unsecured’ network is uncontrolled and carries public traffic on it.

It is possible to set up a firewall between two LAN segments using the IP Office layer 3 switch. IP500 V2 supports a two-port Layer 3 Ethernet switch with the firewall between them. Both of these switched ports have their own IP addresses (LAN1 and LAN2) and in order for traffic to pass from one port to the other, a route is configured in the system’s routing tables.

DHCP Server

IP Office can manage your IP Network for you through its integral DHCP Server. IP Office can be configured to hold a pool of IP addresses for users on the Local Area Network. When a user powers up their PC, the system will allocate them an IP address for the duration of their session. The DHCP server also provides the user's PC with the address of the Domain Name Service (DNS) server and the Windows Name Service (WINS) server. Alternatively, for customers who have a separate DHCP Server, IP Office can be configured to obtain its address from that DHCP server or be set with its own static IP address. Both IP500 and IP500 V2 have two independent DHCP servers, one dedicated to each of the Layer 3 switched LANs.

Leased Line Support

IP Office is capable of connecting to leased line services.

IP Office WAN services are supported over E1/T1 PRI trunks and BRI trunks. E1/T1 trunks can be configured to operate in a fractional mode for 'point to multi-point' applications i.e. a single 2M interface could be treated as 3 x 512K and 8 x 64K going to 11 different locations. When using T1 as a Leased Line it is possible to use the same circuit for switched circuit services. Not all types of leased line are available in all territories, check for availability.

Dial-Up Circuit Support

Where the amount of traffic does not justify the cost of a dedicated leased line, the system can provide data connectivity via ISDN dial-up circuits using its E1/T1 or Basic Rate trunks. Where data speeds greater than a single channel are required (64K/56K), additional channels can be added to the call as and when they are needed.

Point-to-Point Protocol (PPP)

PPP is an industry standard Wide Area Networking Protocol, that allows inter-working with a wide range of 3rd party routers. PPP is used over leased line circuits where a single channel is used to connect the two locations together. e.g. A single channel maybe a 64K channel on a dial-up circuit or a 256K leased line etc.

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Multi-Link Point-to-Point Protocol (ML-PPP)

IP Office supports Multi-Link PPP allowing additional calls to be made where bandwidth greater than a single channel is required. The maximum number of channels available to data can be set on a service-by-service basis. When the available bandwidth reaches a user defined limit additional channels can be automatically added. Similarly, when traffic falls then the number of channels in use can be automatically reduced. If there is no data traffic on any of the channels in use then all lines can be cleared. Since most carriers have a minimum charge for calls, the period that a channel has to be idle before clearing is configurable. Through these mechanisms call costs can be effectively controlled while ensuring that bandwidth is available as and when it is needed.

Frame Relay

Frame relay is a wide area networking protocol based on the X.25 protocol. Individual network connections are multiplexed over a common medium by the use of Permanent Virtual Circuits (PVC). This allows a single Leased Line to provide connectivity to a number of different locations. Frame relay is currently implemented in IP Office as a CPE or 'router end' protocol over WAN connections. IP Office supports both PPP and RFC1490 encapsulation with fragmentation of large data packets to provide voice quality of service.

Service Quotas

IP Office can be configured to limit the maximum number of minutes that a service, such as Internet Access, is available for each user. This is the sum total of calls made and does not include periods of inactivity. Once the quota has been used the service is no longer available. The quota can be either automatically refreshed daily, weekly or monthly or manually refreshed by dialing a secure feature code on a handset.

Time Profiles

Time profiles set the operational times for service. For example, a time profile could be set up to make Internet Access available to staff only during lunch times. Using time profiles it is also possible to define an alternative service to operate outside the operational hours of the main service. This may be used to take advantage of alternative tariffs at off peak periods. Switching to this fallback service can also be controlled manually by dialing a secure short code from a handset. This can be particularly useful in allowing quick restoration of service in the event of an ISP failure. This feature also applies to days of the week or specific calendar dates.

Password Authentication Protocol (PAP)

PAP is a method of authenticating the remote end of a connection using unencrypted passwords.

Challenge Handshake Authentication Protocol (CHAP)

Challenge Handshake Authentication Protocol allows an incoming data call to be authenticated using encrypted passwords. The system also provides the option to periodically reaffirm the authenticity of the caller during the data call.

Data Header Compression

IP Header Compression (IPHC) reduces the header size of the data packet to gain bandwidth efficiency over Wide Area Networks, but adds to transmission latency.

Data Compression

IP Office supports both Microsoft Point to Point Compression and Stac Lemple Ziv to provide greater throughput on slow speed wide area network links.

Bandwidth Allocation Control Protocol (BACP)

Bandwidth Allocation Control Protocol allows the negotiation with the remote end of the data call to request additional calls to be made to improve aggregate data throughput.

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Callback

Three types of call back are supported

- LCP (Link Control Protocol)
  After authentication the incoming call is dropped and an outgoing call is made to a predefined number to re-establish the link.
- Callback CP (Microsoft's Callback Control Protocol)
  After authentication from both ends, the incoming call is dropped and an outgoing call to a predefined number made to re-establish the link.
- Extended CBCP (Extended Callback Control Protocol)
  Similar to Callback CP however, the Microsoft application at the remote end will prompt for a telephone number. An outgoing call will then be made to that number to re-establish the link.

Domain Name Service (DNS) Proxy

Domain Name Service servers provide the translation of names such as www.avaya.com to the domain's IP address required to establish a connection. IP Office provides this service to PCs on the network by proxy.

Network Address Translation (NAT)

Network Address Translation is a mechanism that allows you to use different IP address on your private network behind a router with a public IP Address. When connecting to the Internet, ISPs typically want a customer to use an IP address they have allocated. Using NAT this is easily accommodated, eradicating the need for the customer to change their network numbering scheme and providing additional security to the internal users as their address in hidden to the public.

Typically, a company maps its internal network addresses to a global external IP address and unmaps the global IP address on incoming packets back into internal IP addresses. This helps ensure security since each outgoing or incoming request must go through a translation process. This also offers the opportunity to qualify or authenticate the request or match it to a previous request. NAT also conserves the number of global IP addresses that a company needs.

Proxy Address Resolution Protocol (ARP)

Support for Proxy Address Resolution Protocol allows IP Office to respond on behalf of the IP address of a device connected to it when receiving an ARP request.

Auto Connect

If a service is idle, that is no one is using the Internet, Auto Connect allows the IP Office to periodically connect to a service. This is ideal for mail polling to retrieve email from an Internet Service Provider. An 'Auto Connect Time Profile' controls the time period during which automatic calls are made, for example not at weekends or during the middle of the night.

Firewall

IP Office integrated firewall provides packet filtering of the most common IP protocols including File Transfer Protocol (FTP) and Internet browsing (HTTP). Each protocol passing through the firewall can be restricted/ allowed access in four different ways:

- Drop
  No sessions via this protocol will be allowed through the wall
- In
  An incoming session can "punch a hole" in the wall to allow traffic in both directions
- Out
  An outgoing session can "punch a hole" in the wall to allow traffic in both directions
- Bothway
  An incoming or outgoing sessions can "punch a hole" in the wall to allow traffic in both directions.

In cases where a protocol is not supported by default, the firewall can be customized to control packets based on their content.

IP Office allows the configuration of as many firewalls as needed through IP Office Manager. This permits different security regulations to be applied to individual dial-in users and data services.

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Light-Weight Directory Access Protocol (LDAP)

IP Office supports LDAP directory synchronization. This allows the telephone number Directory (names and telephone numbers) held in IP Office to be synchronized with the information on an LDAP server (limited to 5000 entries). Although targeted for interoperation with 'Windows 2000 Server Active Directory', the feature is sufficiently configurable to interoperate with any server that supports LDAP version 2 or higher.

Remote Access Server (RAS)

IP Office provides RAS functionality allowing external users to dial in to the local area network from modems, telephone adaptors and routers. Several of the previously described features and services can be applied to the dial-in users to create a powerful Remote Access Server. Dial-in users can be authenticated using either PAP or CHAP. Once authenticated the DHCP server can automatically assign the user an IP address to use while connected to the LAN. Individual time profiles and firewalls can be applied to the user restricting what they have access to and when they have access. For further security and accounting ease, IP Office can automatically call a user back. This keeps the cost of the telephone call on the company telephone bill removing the need to process individual expense claims.

Routing Information Protocol (RIP)

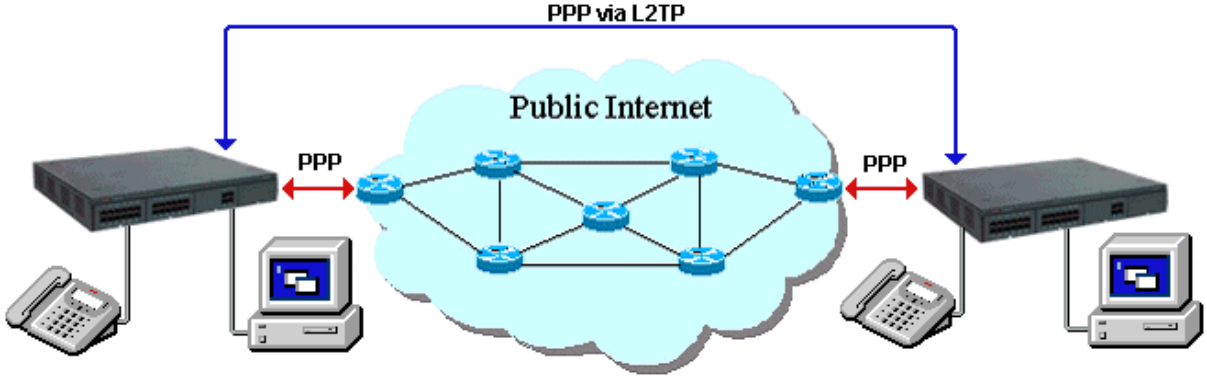
RIP is a distance vector protocol that allows routers to determine the shortest route to a destination network. It does this by measuring the number of intermediary routers that need to be traversed to reach the destination network. If more that one route exists to the same destination the shortest route is used. If a fault occurs on the shortest route it will be remarked as being infinite and any alternative route will become the new shortest route. This behavior can be used to add resilience into a data network. Where a customer has an existing data network comprising of third party routers, IP Office added to the network can provide back up using its routing and dial-up capability. RIP enabled routers share their knowledge of the network with each other by advertising and listening to routing table changes. IP Office Supports both the RIP I and RIP II standards.

VPN: IPSec Tunneling

IPSec tunnels allow a company to pass data between locations over unsecured IP networks such as the public internet. The company data is secured using 3DES encryption making it unintelligible to other parties that might be 'eaves dropping' on the traffic. Tunneling can be applied to link offices together or provide workers access to the office over the internet. All IP Office systems support up to a total of 256K worth of encrypted traffic to multiple locations. Initially, inter-working is supported only between IP Offices that are connected either directly on a WAN port or via the LAN using a 3rd Party router. IPSec is optional and enabled on IP Office through a License Key.

Note: Check with Avaya for supported scenarios and 3rd party devices.

VPN: Layer 2 Tunneling Protocol



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PPP authentication using PAP or CHAP takes place between directly connected routers only. When using a public IP Network to connect sites this authentication takes place between the customers router and the service provide router that it is connected to. In some circumstances it is desirable to authenticate between the customer owned routers, jumping over all the intermediary routers of the service provide network. Layer 2 Tunneling Protocol allow this to happen by facilitating a two stage authentication, firstly with the service provider router then the customer router on the remote network.

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Chapter 10. User Call Control Applications

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10. User Call Control Applications

Avaya provides the following call control application with IP Office Release 6.1:

- Avaya one-X™ Portal for IP Office 274
  Avaya one-X™ Portal for IP Office is a server based applications that the user accesses with their web browser. It can only be purchased as part of the IP Office Office Worker, Power User and Teleworker user licenses.

one-X Portal for IP Office

The Avaya one-X™ Portal for IP Office is an application that provides users control of their telephone from a networked PC.

Avaya one-X™ Portal for IP Office can be used with any IP Office extension; analog, digital or any IP telephones, wired or wireless, and is available as part of the IP Office Office Worker, Power User or Teleworker user licenses only.

Avaya one-X™ Portal for IP Office is a server based application that the user accesses via web browser.

Via separate gadgets, Avaya one-X™ Portal for IP Office provides easy access to telephony features, call information, call and conference control, instant messaging, directory and VoiceMail Pro mailbox.

Calls Gadget

Caller ID/Name Presentation

Caller ID is presented as standard (where provided) allowing users to see who's calling before answering. The caller's phone number and name (if known to IP Office) are clearly shown in the call status area.

The same information is also displayed should a second incoming call be presented, allowing users to easily switch between calls.

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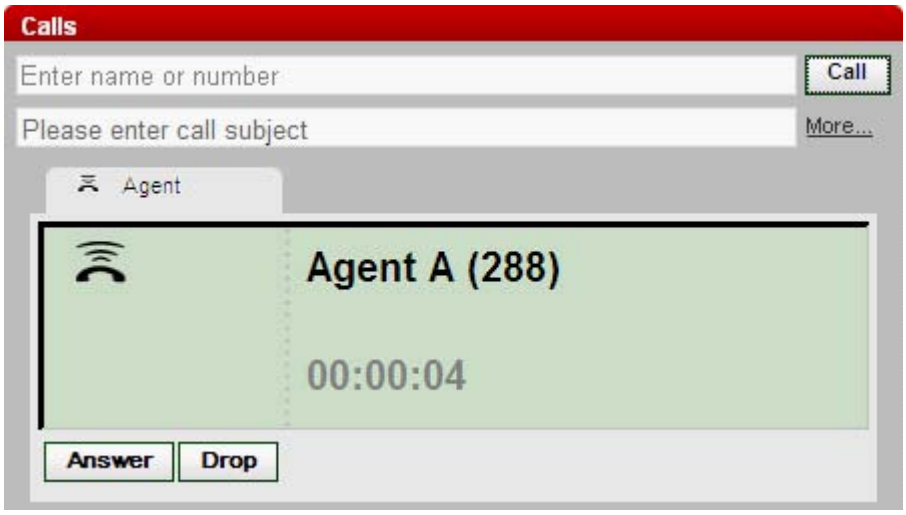

Desktop PC Telephony Controls

Avaya one-X™ Portal for IP Office has telephony buttons in the call gadget that activate standard telephone functions such as Answer, Drop, Hold, Retrieve, Record, Consult and Transfer so that users don't need to remember IP Office specific feature codes. These functions are context sensitive and appear depending on the status of the call. Keyboard shortcuts are available for Answer, Hold, Drop and Call functions and can be configured by the user.

Active calls can be easily parked by clicking on a park slot whilst displaying the active call. Four Call Park slots/zones, which can be shared between users and operators, or within a department on the same IP Office system, further add to the ease with which the entire call handling process is streamlined with Avaya one-X™ Portal for IP Office. Such parked calls can be retrieved either through Avaya one-X™ Portal for IP Office, Phone Manager, SoftConsole or a desktop phone. The call park slot names can be configured by the user.

Conferencing Controls

one-X Portal for IP Office allows the setting up of an ad-hoc conference call from calls on hold and/or the directory, or a meet-me conference. It is possible to configure the Conference ID of the user for meet-me conferences.

Once the conference is set up the conference originator will have the following functions:

- mute/unmute a party/oneself
- mute/unmute all
- drop a party from the conference
- hold the conference while others keep talking
- record the conference

Call Log Gadget

Call History

The call log displays details of calls you have made, received and missed and it will report the last 30 calls. Users can use the call log to make a call or add the caller to the Personal Directory.

The call log shows the actual call history, independent of whether the user was logged in at the time or not. The call log is centralized and also available on the desktop phone.

Calls are ordered in 4 tabs:

- All: all calls.
- Incoming
- Outgoing
- Missed

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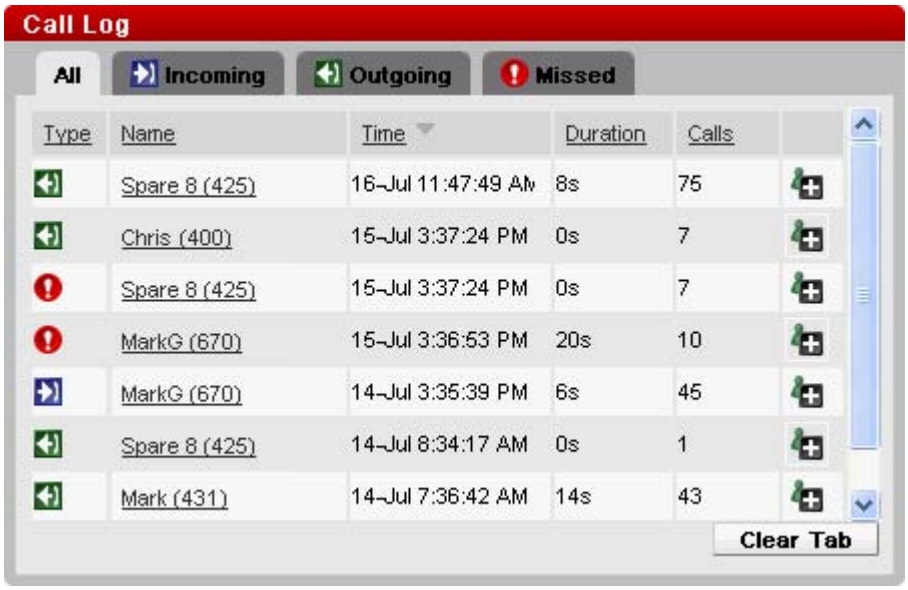


Each tab can be ordered by name, number, time of the call, duration, number of calls and call type. The order can be ascending or descending.

To make a call from the call log, the user simply clicks on the number they wish to call.

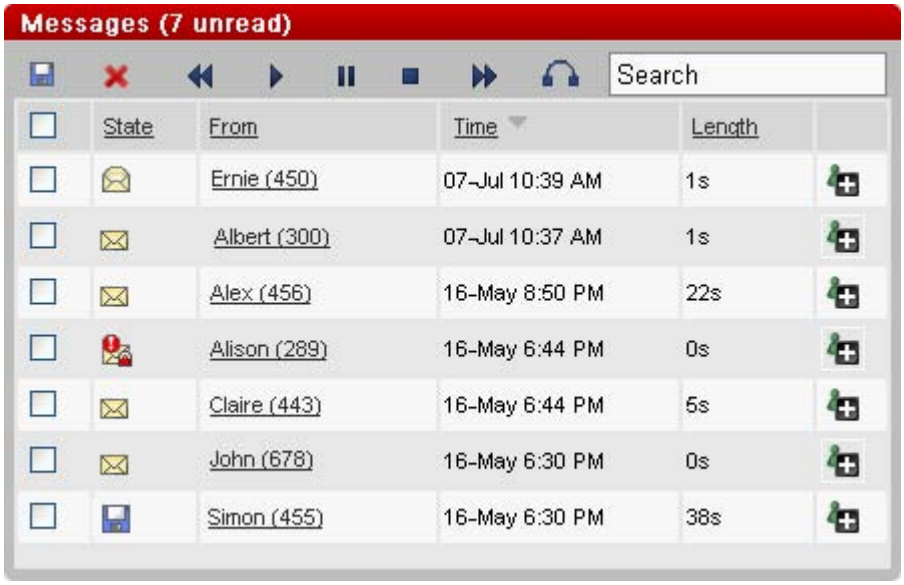
Messages Gadget

Voicemail Access

Avaya one-X™ Portal for IP Office will show new, saved and old voicemails received and provides access into the user mailbox allowing the user to play, rewind, fast-forward, save and delete messages.

The voicemails can be ordered by state (new, saved, read, private, urgent), caller, called party, time and length. The order can be ascending or descending.

The user can easily call back the caller who left a message from the message gadget.



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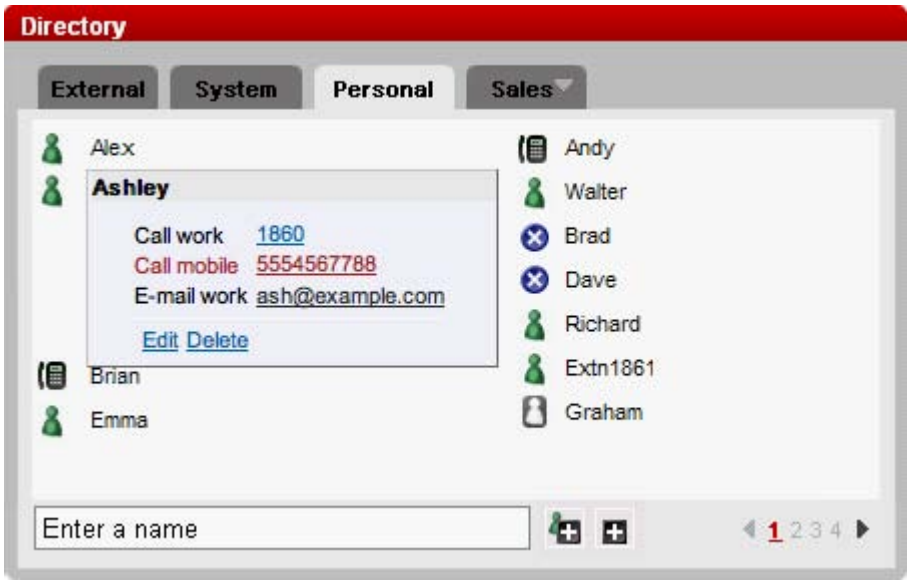
Directories

Avaya one-X™ Portal for IP Office can display several directories of names and associated telephone numbers.



- Personal Directory
  This is the user's own directory of names and numbers. The user can associate multiple numbers (e.g. work, home, mobile, etc.) with a name and select which number to use when making a call. They can also edit and change the directory entries. The personal directory can contain up to 100 entries. The personal directory is common with the 1400, 1600 and 9600 Series telephones (as well as T3 telephones in selected EMEA countries).
  - The Avaya one-X™ Portal for IP Office user can add up to 4 additional tabs for personal directory contacts,
- System Directory
  This is the directory of names and numbers from the IP Office telephone system plus all the users and groups on the telephone system. These entries cannot be changed, but the user can copy a system directory entry into your personal directory and modify it.
- External Directory
  The system administrator can configure Avaya one-X™ Portal for IP Office to access one external directory (Active Directory / LDAP).

Searching the directories is easy: with the Personal Directory and the System Directory, as you enter a name or number in the box at the bottom of the tab, only matching directory entries will remain in view.

With the External Directory, you must enter a name or number to perform a search.

The user can make a call directly from the directory gadget by clicking on the number associated with the entry. If more than one number is available for this entry, they can choose which one to call.

User Status/Presence

For the directory entries of other IP Office users, Avaya one-X™ Portal for IP Office will indicate the status of the user at their work number. This status is available for users across an IP Office Small Community Network.

The status can be:

State	Icon	Description
Available		The normal state for a user when their extension is not in use.
Busy		The normal state for a user when their extension is on a call.
Do Not Disturb		The user has set Do Not Disturb. Calls to them will go to voicemail if enabled or else get busy tone unless you are in the user's Do Not Disturb exception list.
Logged Out		The user has logged out from their phone. Calls to them will most likely go to voicemail if available.
Other		This icon is used when the status is not known.

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Presence within Avaya one-X™ Portal for IP Office allows the user to create sets of call redirection settings. Through the configuration tab the user can create different Presence entries and associate different rules to each such as forwarding to a different number or Do Not Disturb. In case of Do Not Disturb, the user can establish a list of numbers (DND exceptions) from whom the user wants to receive calls.

World Clock Gadget

The World Clock gadget shows you the current time in different time zones previously selected.

Agent Control Gadget

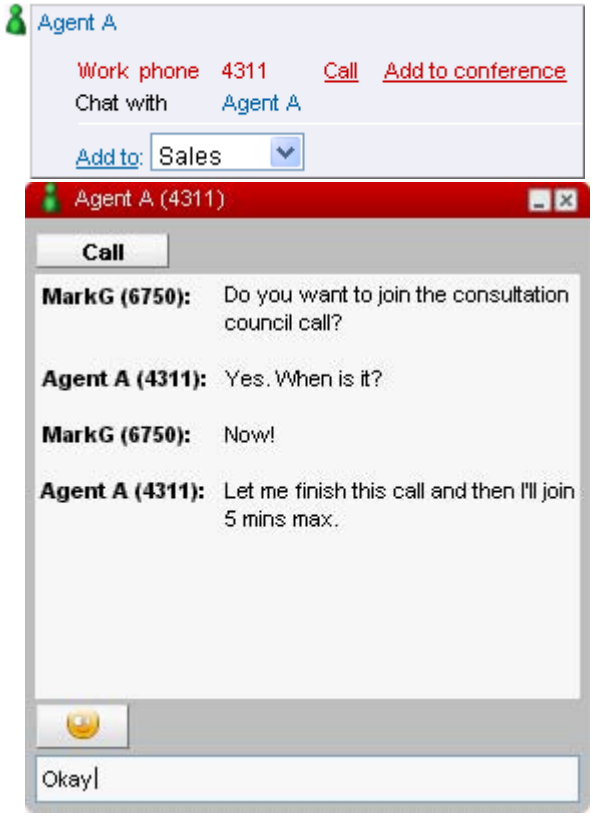
If the user is also a Customer Call Reporter agent, this gadget is displayed. Users can use it to see their current agent state and change that state.

In Agent Control Gadget, Agents are offered the ability to set their state as:
Available
After Call Work
Busy Not Available

It can also be used to change the membership status in the different Customer Call Reporter queues to which the user belongs.

Instant Messaging Another Avaya one-X™ Portal for IP Office User

From the Directory entry of a user the option chat with allows the establishment of an instant messaging session. Such sessions can be established only with other logged on one-X Portal users and can be established while on multiple calls.


Agent A
Work phone 4311 Call Add to conference
Chat with Agent A
Add to: Sales
Agent A (4311)
Call
MarkG (6750): Do you want to join the consultation council call?
Agent A (4311): Yes. When is it?
MarkG (6750): Now!
Agent A (4311): Let me finish this call and then I'll join 5 mins max.
Okay

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Personalizing the Layout

There are a number of ways in which the one-X Portal for IP Office gadgets layout can be adjusted. Any changes made are stored by the one-X Portal for IP Office in your user settings so the saved version is always available when the user logs in again.

Change the one-X Portal for IP Office Layout
· Select a Skin – there are 2 skins available to choose from
· Change the Column Widths – the screen is in divided in two columns of variable width.
· Show/Hide a Gadget – it is possible to minimize unused gadgets and use the available space for the used ones.
· Resize a Gadget – by dragging the resize icon at the bottom of the gadget it is possible to change size to a specific Gadget.
· Move a Gadget – by dragging the gadget itself
· Default the Layout – go back to the factory default.

Profiles

By setting and creating using profiles the user is able to easily control where your calls are directed and how they are treated.

There are 3 default profiles, called Office, Mobile and Home. These can be edited but not deleted. The user can create up to 5 additional profiles. A special profile called Detected may also appear and it is displayed when the current telephone system settings don't match any of the profiles.

By selecting which of the saved profiles is the current one, the user applies that set of settings to the telephone system.

When creating or editing a profile, there are 4 settings that the user will have to select:

- Mobility - The user can choose between:
  - The default desk phone
  - A number to forward the calls to
  - Mobile Twinning with the number to twin
  - Telecommuter with the number where the user is based
- Voicemail - The user can choose to activate/deactivate the following options
  - Voicemail on call not answered
  - Voicemail ringback
  - Voicemail Outcalling
  - Voicemail to be listened on PC or on phone.
  - Active Greeting – The user can choose which recorded greeting to make active.
- Call Pick Up – on/off

Call Assistant

Call Assistant is an optional desktop application that enables the following functions:

Screen pop when receiving a call.

The screen will show the name (if in the personal or system directory) and number of the caller and the user can anser or drop the call by clicking on the relevant button

Outlook Contact Pop

It will also give the option to automatically pop up the Outlook 2003/2007 Contacts entry for the caller or the called.

Hot Key

The one-X Call Assistant can be used to make a call by dialling a number selected in a Windows application. To do this, simply select or highlight the number in the Windows application and then press your one-X Call Assistant's hot key combination.

Feature Summary

Feature	Phone Manager Lite	PMPro	one-X Portal for IP Office
Web browser based	-	-	Yes

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Feature	Phone Manager Lite	PMPro	one-X Portal for IP Office
OS independent (Windows, Linux, Mac support)	-	-	Yes
Click to dial	-	-	Future
Citrix/terminal server support	-	-	Future
Inbound/outbound call handling	Yes	Yes	Yes
Phone call control	Yes	Yes	Yes
Configure phone preferences	Yes	Yes	Yes
Configure keyboard short cuts	Yes	Yes	Yes
CLI (ANI) / Name display	Yes	Yes	Yes
Speed dial / Busy Lamp Field management	Yes - 15 icons maximum.	Yes - 100 icons maximum per tab.	Yes - from Personal & System Directory
Speed Dial tabs (to group Busy Lamp Field icons)	Yes - 1 tab.	Yes - 10 tabs maximum.	Yes – 5 Personal Directory tabs maximum.
Transfer call by drag and drop to a speed dial icon	–	Yes	-
View internal users' presence	Yes (Telephony + LCS[1])	Yes (Telephony + LCS[1])	Yes (Telephony + User Status)
Send Instant Messages (IM) to internal users	Yes (requires LCS[1])	Yes (requires LCS[1])	Yes
Telecommuter mode	–	Yes (not PC SoftPhone)	Yes
Mobile Twinning Configuration	-	-	Yes
Call Pick Up	-	-	Yes
Compact mode	–	Yes	Yes – Call Assistant
Personal Phone Directory	–	Yes - 1000 entries maximum.	Yes – 100 entries maximum
System Phone Directory	–	Yes - 5000 entries maximum.	Yes – 5000 entries maximum
External Phone Directory (e.g. LDAP/Active Directory)	–	Yes - 5000 entries maximum.	Yes - unlimited
Call history log – all, missed, messages.	Yes – while logged in.	Yes– while logged in.	Yes - 24/7 call log
Separated incoming/outgoing call log.	–	Yes	Yes
Collect new voicemail messages.	Yes	Yes	Yes
Voicemail box control (Intuity and IP Office modes).	–	Yes	Yes
Personal Distribution List set up (Intuity mode)	–	Yes	-
Incoming call scripting.	–	Yes	-
Time on call.	–	Yes	Yes
Advice of Charge (ISDN service provider dependent)	Yes	Yes	-
Door opening control.	–	Yes	Yes (via analog extension)
Queue monitoring.	–	Yes - 2 Queues	Future
Conference Control Display.	Yes	Yes	Yes
Audio Conference Management	No	No	Yes
'Screen pop' contacts with Outlook	–	Yes	Yes
Simple Outlook contact record creation.	–	Yes	-
Agent Mode.	–	Yes	Yes
Distinctive Ringing (WAV file).	–	Yes	-
Post Connect dial (sending DTMF while connected to another party).	Yes	Yes	-
VoIP mode (to run as an PC Softphone)	–	Optional license	Yes (with Power User and Teleworker licenses only)

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Feature	Phone Manager Lite	PMPro	one-X Portal for IP Office
Video Softphone	-	-	Yes (with Power User and Teleworker licenses only)

1. LCS: Microsoft Live Communications Server 2003

Server, Browser and Other Requirements

- Phone support
  Any telephone connected to IP Office, although handsfree operation is only supported on suitable Avaya Digital and IP telephones.
- Server and Browser requirements
  - Avaya one-X™ Portal for IP Office supports the following browsers:
    - Microsoft Internet Explorer versions 7 and 8.
    - Mozilla Firefox version 3.0 and above.
    - Windows Safari and Apple Safari 3.1 and above.
  - The server requirements are:
    - Intel Pentium D945 core/AMD Athlon 64 4000+; with 3 GB RAM and 20 GB free hard disk space.
    - OS supported: Windows Server 2003 (32 and 64 bit).
    - Always refer to the latest Avaya IP Office Technical Tip or Technical Bulletin for any updated information with regard to Operating Systems, Service Packs or PC hardware.
- Licensing
  Avaya one-X™ Portal for IP Office requires a Avaya one-X™ Portal for IP Office license for each user. This license is part of the IP Office Power User, Office Worker or Teleworker user licenses and is not available separately. The Telecommuter mode and SoftPhone are available only with Power User and Teleworker. Softphone does require an Avaya IP Phone license on IP Office and it is part of profile licensing.

IP Office Video Softphone

With Release 6 IP Office introduces – as part of the Teleworker and Power User package – a new Softphone application.

With Release 7 a number of new features have been added to Video Softphone:

- Busy Lamp Field allowing to see telephone status of colleagues including pickup of alerting calls
- HD Video support, delivering high quality Video conferences between Softphone users as well as with other supported Video endpoints.

The Softphone is a full feature telephony client that supports standard telephony features on a PC running supported Microsoft operating systems including new Microsoft Windows 7.

Support of several audio options including support of wireless headsets with full hook-switch support provides a flexible telephone option, especially for people on the move.

Softphone is part of the Teleworker and the Power user package. An applicable Teleworker/ Power User license for the user of the Softphone is required; however Softphone does not require any additional license, e.g. no (Avaya) IP-station license.

Softphone can run in two different operating modes:

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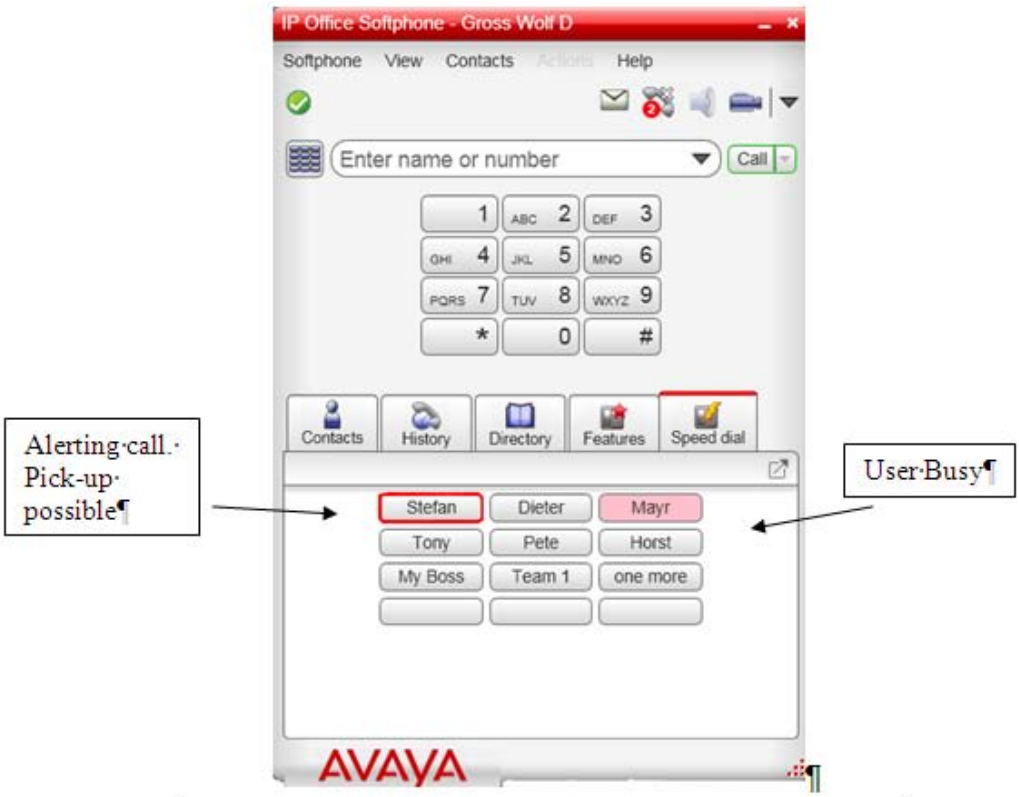
a) Softphone client mode: The Softphone provides its own User interface, similar to a standard IP Office telephone for making calls, transferring calls, multiple call appearances or making conference calls. A programmable feature button list allows one-touch access to standard IP Office features. This is the preferred usage scenario for customers not running other IP Office productivity applications like one-X portal. It is also the preferred operation mode for users who want to smoothly migrate from their desktop phone or who use Softphone only occasionally and prefer a phone-centric user-interface, with optional dial pad and standard phone feature keys. The user interface is flexible, allowing adjustment for personal preferences with detachable specific windows like the directory and call log window. The attached picture shows the Softphone with dial pad and feature keys, plus a view of the contacts of Softphone.



Softphone with new Speed-Dial/Busy-Lamp-Window

Softphone can also be configured to be minimized to the system tray when not in use to smaller or larger configurations to save desk-space.

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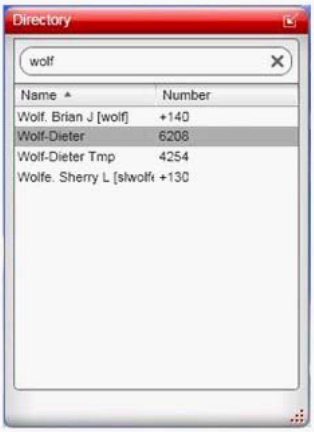
IP Office Softphone with dialpad, active call and Call Log (History)

Deskphone buttons for IP Office Features



IP Office Directory Sample

Video Picture: Active Video Call with self-view

IP Office Softphone in smallest mode

b) Application mode: the Softphone runs in the background and standard call control is accomplished through productivity applications, such as one-X Portal for IP Office that is part of the User profiles described above.
If run with one-X portal it provides the following additional value:

1. Consistent user interface when using one-X Portal with a desktop phone, in telecommuter mode or in combination with Softphone.
2. Off-line call log with updates even when Softphone is not logged in
3. Additional features of one-X Portal e.g. presence and instant messaging.

Video Support

As well as audio calls, IP Office Video Softphone supports end to end video calls on a single IP Office or within an IP Office Small Community Network. Currently, video is limited to end to end calls, no multi-party conferencing is supported.

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Video adds a new level of interaction to the standard phone communication so this is especially useful for users working from a remote location, needing to get in touch with colleagues in the main site or when used in an IP Office Small Community Network.

Feature Overview

The following feature overview is applicable when using the Softphone in Client mode. When used in application mode, functionality depends on that application.

- Making and receiving calls
- Multiple Call Appearances (No bridged or line appearances)
- Hold, transfer (attended and unattended)
- Message waiting with direct voicemail access
- Audio Conference (IP Office based)
- Local call log, local directory
- Access to IP Office directory (system, corporate, personal)
- Feature keys for up to 12 IP Office features like:
  - Forwarding (various methods like all, busy, no answer)
  - Park/Unpark
  - Pick up (any, group, member)
  - Voice-Mail on/off
- Speed Dial Tab with Busy Lamp Status:
  - Free Configurable Speed Dials: 12-60 Buttons
  - Visual Indication of User Status (Busy, Alerting)
  - Pick-Up of Alerting calls

Features keys can be selected and self administered from a list of features that is downloaded from IP Office.

If equipped with an audio device with hook switch control, Softphone allows calls to be made or answered even when the screen-saver of a PC is activated and the PC is locked. Until the PC is unlocked, no screen will be shown but ringing calls can be accepted through e.g. a headset with hook-switch.

Licensing and Provisioning

IP Office Video Softphone forms part of the Teleworker or Power User profile. A user can log onto the Softphone using their normal username and password. License entitlement will be checked against that user before the login is accepted.

Unlike Avaya hard telephones, Softphone does not require an Avaya IP Phone License to work on IP Office.

While the Softphone is a thick client application, it is extremely easy to install and provision. Installation of the application on a standard PC requires minimal to no configuration as all provisioning of the product is done from the IP Office as a provisioning server.

If a DHCP server is present and set up correctly, Softphone will completely configure itself. The user only needs to enter the username and password to get connected.

If DHCP is not available, the user needs to enter a single IP address of the IP Office; all other parameters will be set automatically.

Softphone can be set up so that, when the PC starts up, the application is automatically started with the default username and password. The user can disable automatic login as well as interrupt it any time.

Username and Twinning

When logged in to Softphone with their username and password, the user will automatically be logged off from their existing desktop extension, so the user is fully available on their normal number without any additional steps.

For users that are using a physical telephone e.g. in the office, this is very convenient as they do not use a different extension for Softphone calls. All calls will be routed automatically to the Softphone when logged in. There is no need to use twinning to tie the Softphone to the standard desktop phone.

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Supported Audio and Video Hardware

Softphone can run with any Windows sound device supporting full-duplex quality.

If possible, audio devices are automatically detected and allocated to the Softphone.

Softphone supports simultaneous operation of two sound devices, e.g. a headset or USB handset for normal calls and a USB handsfree unit for handsfree operation. It is possible to switch between the sound devices at any time during the call.

For alerting, in addition to that the PC speakers can be activated to ring the phone.

For ease of use, similar to desktop telephones, Softphone supports USB devices for audio as well as for call control, e.g. accepting and terminating a call. Softphone supports the “HID” standard for USB devices allowing generic USB devices to be configured.

A number of USB devices have been tested with Softphone for interoperability, including Hook switch control as well as dialing (if supported by that device).

- Audio Devices:
  - Plantronics DSP (USB headset)
  - Plantronics 510 (Bluetooth headset with USB base)
  - Plantronics CS50 (USB wireless headset)
  - GN Netcom 8120 (USB headset adapter for GN Netcom headsets)
  - Yealink W1DL (Cordless USB phone with Dialpad)
  - Yealink P1K (Corded USB Phone with Dialpad)
  - Generic USB soundcard and USB HID interface support (with configuration assistant)
- Video Cameras:
  - Logitech QuickCam Pro 4000, 5000, 9000
  - Creative Webcam NX Pro
  - Microsoft Lifecam VX-3000
- Video Cameras for HD Video.
  - Logitech Webcam Pro 9000:

Supported Audio and Video Codecs

Softphone supports G.711 and G.729 Audio Codecs. Audio Codec negotiation is done by IP Office in order to ensure best quality for local or SCN calls. At login, the user can select the default mode, e.g. best quality or low bandwidth for remote operation.

For Video, Softphone supports H.263, H.263+ and H.264 Video Codecs with resolution of up to 1280*768 pixels and 30 frames per second.

Resolution	Pixel	Typical Net Video Bandwidth Requirement	
		H.263	H.264
Low	176 x 144 (QCIF)	64 kbps	45 kbps
Standard	320 x 200 (QVGA)	162 kbps	114 kbps
High	640 x 480 (VGA)	776 kbps	545 kps

Note: The above numbers show an average bandwidth requirement but peak usage might be higher. The above numbers do not include the audio-bandwidth.

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Supported Languages

Currently the following languages are supported:

- English,
- French,
- Spanish,
- Italian,
- Portuguese
- Simplified Chinese

Over the time, new languages can be added to the product.

PC Requirements

- Processor:
    - Minimum: Pentium 4 2.4 GHz or equivalent, Video Card with DirectX 9.0c support
    - Optimal: Intel Core 2 Duo or equivalent, Video Card with DirectX 9.0c support.
- Memory:
    - Minimum: 1 GB RAM
    - Optimal: 2 GB RAM
- Hard Disk Space: 50 MB
- Operating System: Windows XP Service Pack 2, Windows Vista, 32-bit and 64-bit, Windows 7 32-bit and 64-bit.
- Additional: Microsoft Windows Installer 3.1, Microsoft .NET 3.5 SP1, Microsoft VC 9.0 Runtime Service Pack 1
- Connection: IP network connection (broadband, LAN, wireless)
- Sound Card: Full-duplex, 16-bit or use USB headset

For HD video, the minimum requirements are Intel Core 2 Duo 2.3 GHz or equivalant, hardware video acceleration, camera and 2GB RAM

Note that these are recommendations. Depending on other applications on the PC, Softphone cannot guarantee uninterrupted Voice and/or Video quality. While Softphone runs with high priority on the PC in order not to be impacted by normal applications other programs with similar priority could impact Softphone performance. Example applications could be other Multimedia or communication applications like web-sharing programs or programs that are essential for PC performance like some Virus-scanners or firewall programs.

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Chapter 11. Receptionist

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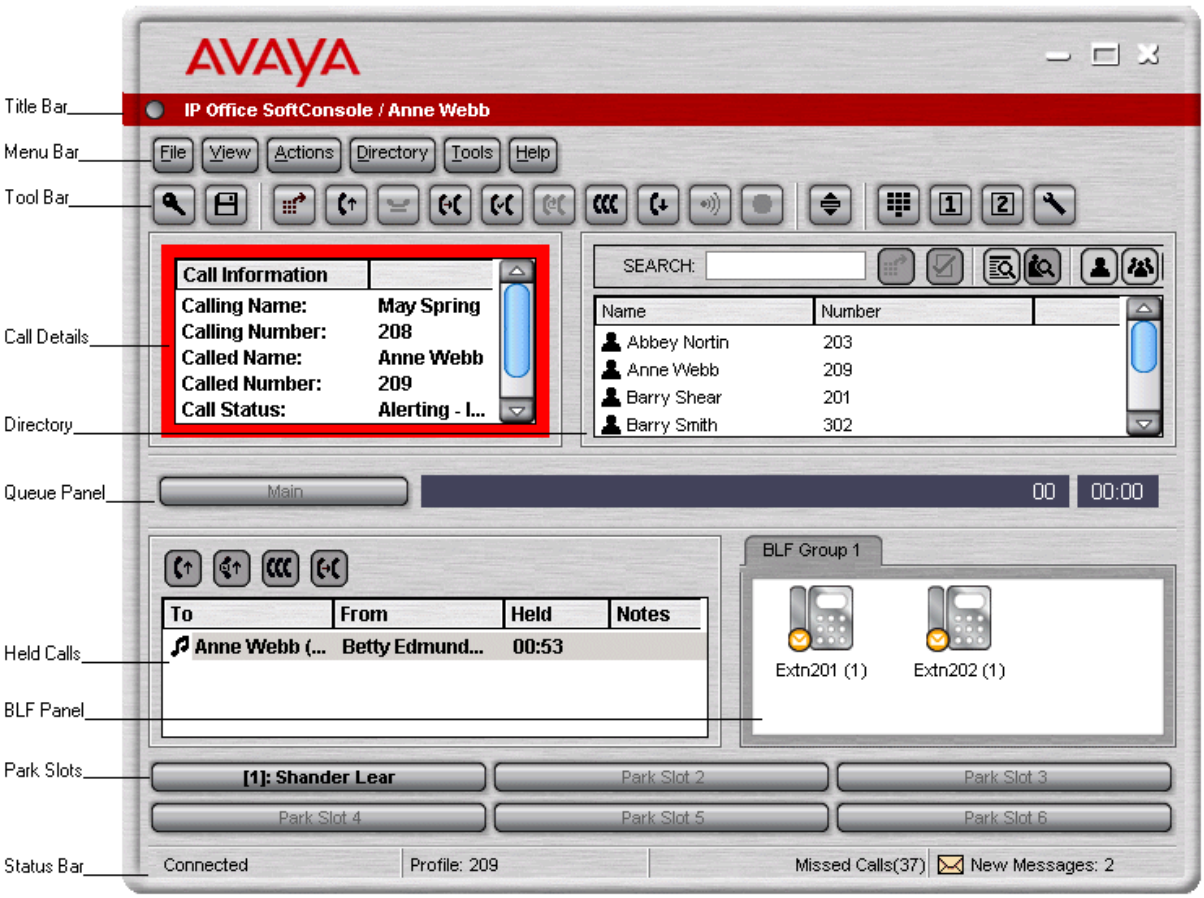
11. Receptionist

SoftConsole is the PC based Windows Receptionist solution for IP Office. It can be purchased with the Receptionist user license.

SoftConsole has been designed to improve operator service by providing the operator with call information and available call actions to simplify call handling and give the appropriate response to the caller. With this easy to use software tool the operator can maintain visibility of the number and type of calls waiting and so ensure that clients are greeted in a professional manner. SoftConsole can be minimized in the Windows system tray when not in use, but will pop up on the screen when a call is received.



SoftConsole has been designed to be easy to use, while offering a look and feel which will appeal to experienced and novice operators alike.

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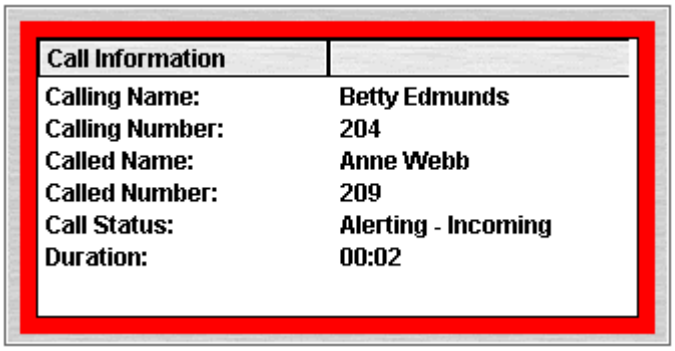
The SoftConsole screen is divided into the following areas:

- Main Menu Bar

Commands and actions are available through menus. Some features can only be used when the right conditions. If they are not available, the feature will be "grayed out" until conditions change that allow the feature to be used. The following features are available on the tool bar:

- Login.
- Save Profile.
- New call.
- Answer call.
- Hold call.
- Transfer call.
- Transfer complete.
- Reattempt transfer.
- Conference.
- Hang up.
- Page.
- Record call.
- Compact view.
- Dial Pad.
- Access conference room 1.
- Access conference room 2.
- Options.

- Call Details Panel



The call details panel on the left shows details of the current call which will include the following information:

- Calling Name
  The system directory name associated with the calling number.
- Calling Number
  The telephone number of the call originator.
- Called Name
  The system user name or hunt group name associated with the called number.
- Called Number
  The extension number the incoming call has been routed to by the system.
- Call Status
  States the progress of a call. The border around the call status panel changes color to indicate the status of the call.
- Call Duration
  The length of time that the has been in the state as indicated by the Call Status
- Notes
  This area displays notes or information about the call i.e. when a call has been returned as there was no answer from the extension it was transferred to. If annotation is attached to the call, details are shown in the Notes area.

If a new call arrives, the call details panel will display the calls waiting to alert the operator and allow answering of the call based on the Caller ID.

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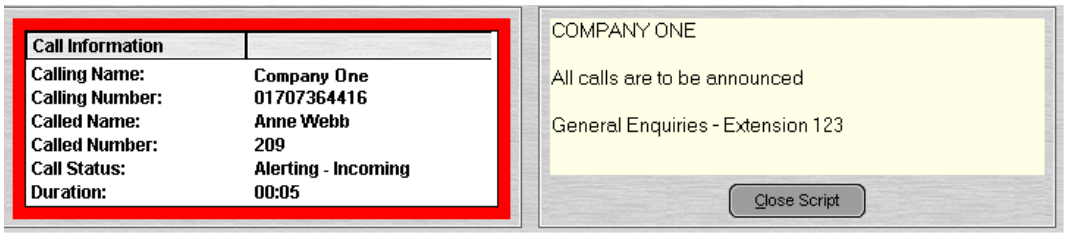
- Directory Panel

Property	Value
Name:	Anne Webb
Number:	209
Busy Status:	Idle
Do Not Disturb Status:	Off
Login Status:	Logged In
Group Status:	
Main	Out of Group
Sales	In Group
CustomerHelp	In Group
Absent Message:	
New Voice Mail Messages:	2
Forwarding Status:	
Forward Unconditional:	Off
Forward On No Answer:	Off
Forward On Busy:	Off
Follow Me:	Off
Forward Hunt Group Calls:	Off

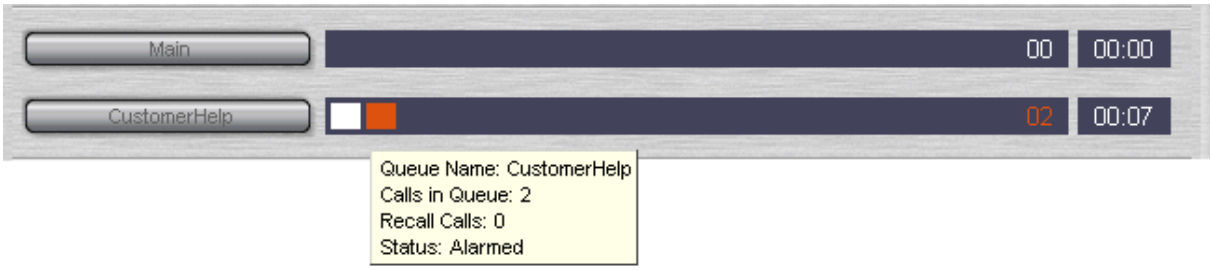
The directory panel on the right shows information on following:

- Directory entries
  Including IP Office users, hunt groups and external directory user (non IP Office extensions)
- Single directory entry details
  Including IP Office users, hunt groups and external directory user (non IP Office user).
- Script
  When a script has been configured for either the calling or called number, the script is displayed in this panel. For example, an operator may be answering calls on behalf of more than one company. To ensure the call is answered with the correct company name a script file can be created with the company name details. The script is displayed whenever a call is received for that company.



  - Conferencing
    Within SoftConsole, calls can be conferenced when held, or a conference can be created through the two conference rooms:
    - Conference Held Calls
      An operator can conference calls that are in the Held Panel. All calls in the Held Panel will be conferenced.

- Queue Panel
  The queue panel displays a bar graph of the number and the status of external calls held in a particular queue. Up to 8 Call queues can be configured and labeled to reflect incoming calls for specific hunt groups.



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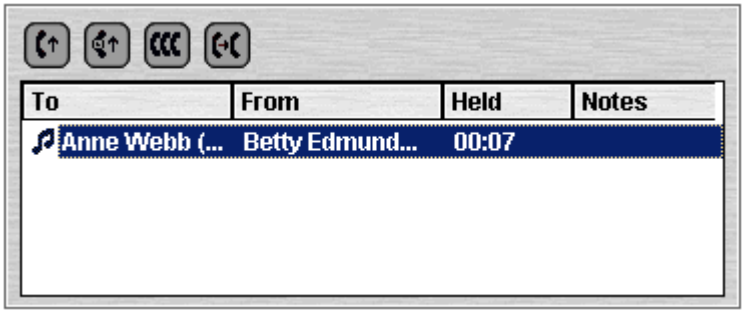
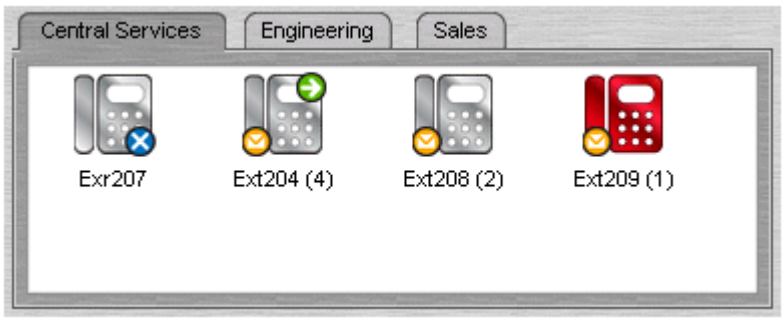
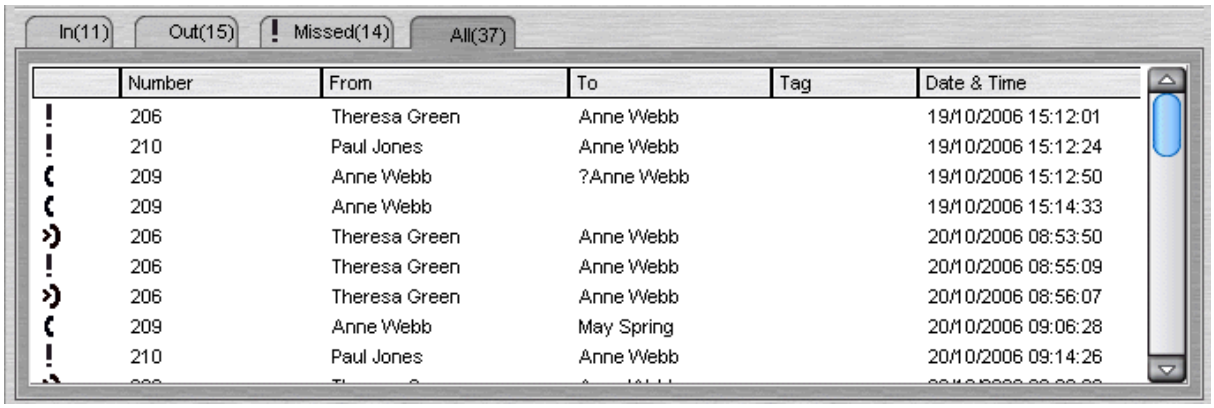
- Held Calls Panel
  The held call panel enables the operator to manage all calls held at the operator station. These calls will appear as a list in panel. The operator can perform the following the functions: Answer the highlighted held call, Answer the longest held call, Conference held calls (see conferencing section above) or Transfer held call.



- BLF Panel (Busy Lamp Field Panel)
  The BLF panel displays icons to indicate the status of selected users. Each icon provides information on individual users such as: Unread 'User' voicemail messages, User status information, for example Busy, DND and Forwarded is indicated by the various icons used. Up to 10 tabs with 100 icons on each tab are supported.



- Park Slot Panel
  The park slot panel can contain up to 16 system-wide park slots with specific Park ID's for each slot.
- Call History
  SoftConsole's call history keeps a combined record of up to 100 (incoming, outgoing and missed) calls while the application is active Double-clicking any logged call dials that number.


In(11)
Out(15)
Missed(14)
All(37)
Number
From
To
Tag
Date & Time
206 Theresa Green Anne Webb 19/10/2006 15:12:01
210 Paul Jones Anne Webb 19/10/2006 15:12:24
209 Anne Webb ?Anne Webb 19/10/2006 15:12:50
209 Anne Webb 19/10/2006 15:14:33
206 Theresa Green Anne Webb 20/10/2006 08:53:50
206 Theresa Green Anne Webb 20/10/2006 08:55:09
206 Theresa Green Anne Webb 20/10/2006 08:56:07
209 Anne Webb May Spring 20/10/2006 09:06:28
210 Paul Jones Anne Webb 20/10/2006 09:14:26

- Status Bar
  This Shows current status of the system and is divided into four sections that display current connection status, current Profile name, information messages and The number of new voice mail messages for the operator. Information messages include any alarm conditions that are present within the system.

SoftConsole Options

SoftConsole has many configurable options available to the operator to personalize the look and feel. The Operator can tailor the usability specifically to each their personal preferences. The following configuration options are available:

- Incoming Calls
  This tab enables the operator to manage the local SoftConsole directory by creating, editing and deleting entries from the selected directory. Also the operator is able to associate a script or media file with each specific entry.

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- Queue Mode
  This tab enables the operator to configure the queue window with up to 8 hunt group queues, which will include a recall queue. Queues can be created, edited and deleted while also providing the operator with the additional benefit of positioning them in the queue window in order of operator preference. Management by exception is used to monitor queue status by enabling the operator to set up various alarm thresholds such as the Number of calls in queue and Longest waiting call time. A WAV media file can be associated with an alarm for further customization.
- Park Slots
  This tab enables the operator to define which park slots are accessible on a system wide basis up to a maximum of 16. The operator is also able to assign which numbers are used to access each park slot and where the slot appears in the park slot panel.
- BLF Groups
  This tab allows the operator to create and edit BLF groups.
- Door Entry
  This tab allows the operator to configure up to two door entries.
- Directories
  This tab enables the operator to choose access to the following directories: SoftConsole local directory, IP Office system directory and Microsoft Outlook contacts. Once chosen, the operator is able to map fields to directory entries.
- Conferencing
  This tab allows the operator to set up the names of the two conference rooms. The names will appear on the telephone displays of users in the conference room (maximum of 10 characters).
- Keyboard Mapping
  This tab allows the operator to assign keyboard short cut keys for SoftConsole functions.
- Keyboard Actions
  This tab allows the operator to specify the default action when alphabetic or numeric characters are pressed.
  - Alphabetic Keystrokes: Begin directory search or Open call annotation window
  - Numeric Keystrokes: Begin directory search or Open pop-up dial pad
- Appearance
  This tab allows the operator to change the appearance of SoftConsole fonts, skins and the call information window color.
- SoftConsole
  This tab allows the operator to save the changes made to the configuration of SoftConsole either automatically or manually to a local configuration file on the PC.

SoftConsole Administration

SoftConsole has an administration mode that enables the operator to configure the following settings:

- Control panel views
  The BLF panel, call history panel, held calls panel and park slot panel can be hidden or made visible.
- Change the Administrator password
- Edit operator profiles
  Each operator can have a personalized profile, which can be configured by the administrator.
- Create and modify templates
  SoftConsole comes with three predefined templates, which can be modified, or new templates can be created.
- Specify the maximum length of call notes
  IP Office supports a wide range of different telephone types. These have different display sizes so the operator can define the character length of messages sent to each user according to the type of phone they use.
- System Tray working
  The application can be minimized and left running in the system tray so that it can pop on received calls.

SoftConsole Telephone Requirements

- SoftConsole provides extensive call management, but it still requires an IP Office telephone to provide the speech path. SoftConsole has been tested and is certified to work with all Avaya wired digital and IP telephones that are listed in "Telephones" 98 chapter.
- SoftConsole cannot be used with IP DECT 3700 Series telephones.

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SoftConsole PC Requirements

- IP Office software release 2.0 or later.
- PC requirements:
  - Always refer to the latest Avaya SME Technical Tip or Technical Bulletin for any updated information with regard to Operating Systems, Service Packs or PC hardware.
  - Refer to Technical Specifications section of the Product Description for Operating System and Hardware requirements.
- A maximum of four SoftConsole applications can be run per system. An IP Office license controls the number of simultaneous SoftConsole users.
- SoftConsole requires a SoftConsole license for each user. This license is part of the IP Office Receptionist user license.

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Chapter 12. Computer Telephony Integration

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12. Computer Telephony Integration

Computer Telephony Integration (CTI) is about bridging the gap between the telephone system and business applications. On IP Office, this is achieved by use of the IP Office CTI Link, a CTI middleware product and Software Developers Kit.

On IP Office, CTI is delivered through adherence to open standards. This gives businesses access to a wide range of third-party solutions, addressing vertical markets, and designed to meet their requirements. For developers, migrating their offering from other platforms to IP Office is quick and easy, and the advanced CTI features IP Office offers makes it easy to demonstrate full integration, and more business benefits.

IP Office provides two levels of CTI interoperability: CTI Link Lite, which is free of charge, provides all the functionality required to support the vast majority of applications, including screen-popping, and many third-party products.

CTI Link Pro provides enhanced functionality, including the ability to control multiple telephones and gives access to advanced call center operation.

Because IP networking is integrated into the IP Office system, all CTI is done through the LAN. This introduces additional points of failure, as well as relying on non-standard interfaces and handsets. On IP Office, all devices can be used with CTI.

CTI with IP Office

IP Office offers a significant CTI capability. Several interfaces are supported:

- TAPILink Lite.
- TAPILink Pro.
- TAPI-WAV driver.
- DevLink Pro.
- IP Office SMDR.
- IP Office Software Development Kit.

- TAPILink Lite
  Provides first-party CTI support for Microsoft TAPI 2.1 and TAPI 3.0, so each PC can control or monitor one phone. The software components are supplied with the IP Office system on the User CD-Rom, and do not required a license key for use.
- TAPILink Pro
  Provides third-party CTI support for TAPI 2.1 and 3.0. These components are identical to their first-party equivalent; the presence of the CTI Link Pro RFA license key (which can be purchased in the usual way for products) enables this additional functionality.
- TAPI-WAV driver
  Provides software-based support for voice processing. The TAPI-WAV driver is for use with TAPI 2.1 only; for TAPI 3.0, IP Office supports the Media Service Provider (MSP) interface, defined by Microsoft in TAPI 3.0. This functionality will only work in conjunction with CTI Link Pro and also requires TAPI WAV licenses. Each license enables 4 ports of voice processing.
- DevLink Pro
  Provides a real-time event stream in addition to the SMDR interface provided in IP Office SMDR. The real-time event stream takes the form of a call record, which is issued whenever the state of any endpoint of a call changes (typically there are two end points on a call, but for some circumstances, such as conference calls, intruded calls there may be more).
- IP Office SMDR
  Provides an interface to obtain SMDR events. A comma-separated record is issued for each call, when the call is completed. This interface is designed for call accounting and call billing applications. The IP Office SMDR can be output direct from the IP Office control unit to a designated IP address and port.
- Software Development Kit
  This toolkit is delivered on a single CD-Rom, containing the developer documentation for TAPILink Lite, TAPILink Pro, DevLink Lite and DevLink pro, as well as pre-compiled programs for exploring TAPI 2.1 and 3.0. In addition, example source code is included, making it easy for developers to become familiar with IP Office CTI interfaces.

TAPILink Lite (1st Party TAPI Support)

TAPILink Lite provides simple first-party CTI via Microsoft TAPI 2.1 and 3.0. Individual desktop PCs connected to the Local Area Network communicate with IP Office via an IP connection over the LAN. Each PC is capable of controlling one telephone device (see diagram below).

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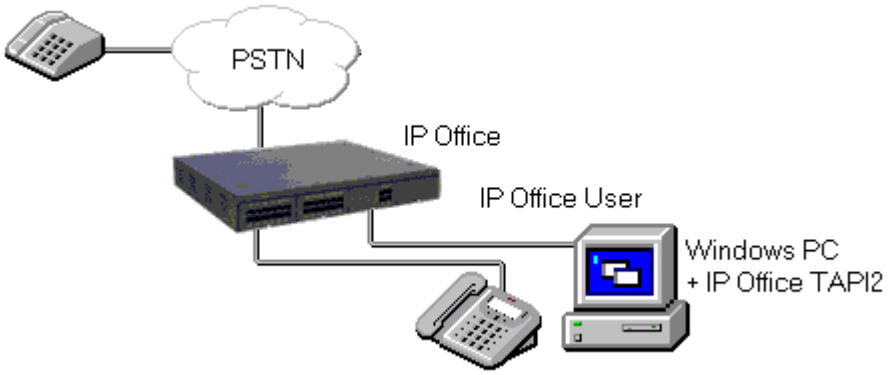

Microsoft TAPI 2.1 and 3.0 are specifications and developers interfaces for controlling and monitoring a telephony device. The specification requires that a certain amount of core functionality is implemented, and additionally defines a series of optional functionality that switch vendors may also implement.

TAPILink Pro (3rd Party TAPI Support)

TAPILink Pro provides all of the features and functionality of TAPILink Lite, but additionally provides third party CTI operation. This means that a single server can control and monitor any number of telephone devices.

In addition, TAPILink Pro provides the ability to monitor and control groups. This allows an application to be notified when a call enters a queue, and can also redirect it to another location.

TAPILink Pro also supports additional TAPI functionality that is not available through TAPILink Lite. This functionality is supported through the LineGetLineDevStatus and LineDevSpecific calls. The additional features are:

- Agent login.
- Agent logout.
- Set and retrieve divert destination.
- Set and retrieve extended divert status (Forward All Calls, Forward on Busy, Forward on No Answer, Do not Disturb).
- Retrieving the extension locale (language).
- Set and clear the message waiting lamp.
- Enable and disable group membership.
- Generate and detect DTMF digits and tones (requires the TAPI-WAV driver).

Support for Developers

The Developer Connection Program ("DevConnect") is the Avaya developer partner program, and is designed for third-party companies who are creating a product for sale, and who wish to receive technical support. Membership of the program is at the sole discretion of Avaya.

DeveloperConnect members pay an annual fee, for which they receive technical support directly from Avaya. In addition, Avaya will perform interoperability testing between IP Office and the member's product, and may also create opportunities for joint marketing, including exhibitions, use of Avaya's logo, and other benefits.

More information on the DeveloperConnect program can be found at www.devconnectprogram.com.

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Chapter 13. Messaging

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13. Messaging

Messaging enables users to manage all of their messages, in particular emails and voicemails, in one place. Since the main messaging platform is typically email, IP Office Preferred Edition enables voicemails to be managed via the email system in order to keep all messages synchronized through one user interface. Essential Edition also enables some basic messaging through the ability to forward voicemail messages to the user's email inbox.

Voicemail in general provides a telephone answering machine with a personalized greeting on every employee's desk and allows callers to leave spoken messages when the user cannot answer a telephone call. Voicemail messages are retrieved either locally or remotely via any telephone (users are prompted for a PIN if they are using any telephone other than their allocated extension or a trusted location e.g. mobile telephone).

The voicemail server is multi-lingual and can offer different prompts depending on the user's preferred language, independently of the default system setup. Similarly, external callers can hear prompts in their own language depending on their incoming call route (e.g. based on caller ID).

Voicemail options available are:

- IP Office Essential Edition:
  - Embedded Voicemail
- IP Office Preferred Edition:
  - VoiceMail Pro – for single site use but use in a Multi-site Networking (SCN) from remote users
  - Distributed VoiceMail Pro – for multi-site use in a Small Community Network (SCN)
  - Centralized INTUITY Audix / Modular Messaging Voicemail – for use with Avaya Communication Manager

  IP Office Preferred Edition is available for Windows and Linux® OS. The Preferred Edition on Linux provide the same functionality as described further in this chapter except of:
  - TTS - The implementation uses SAPI and the Windows versions of the Nuance TTS engines.
  - Use of the MAPI interface for capabilities such as Exchange Integration and visibility of voicemails and faxes in the exchange mail store.
  - Web Voicemail is provided by the user productivity application one-X Portal which comes with Office Worker, Teleworker and Power User licenses.
  - ContactStore integration for call recording
  - VPNM

Positioning Summary

For further details refer to Voicemail Feature Comparison 321 at the end of this section.

Feature	Preferred Edition VoiceMail Pro	Essential Edition Embedded Voicemail
Supported IP Office Systems	IP500 V2 IP500.	IP500 V2 IP500.
Mailboxes	IP Office creates mailboxes for each user and hunt group on the system based on channel licenses.	
Message Storage Capacity	1MB per minute up to hard disk capacity.	Up to 15 hours.
Maximum Simultaneous Calls	Up to 40[1]	IP500 V2 = Up to 6[2] IP500 = 4
Resilience and Backup	Yes	No
Small Community Network (SCN) centralized operation	Yes	No
Distributed Voicemails in an SCN	Yes	No
Voicemail to email forwarding	Yes	Yes
Unified Messaging (UMS)	Yes	No
Microsoft Exchange Server 2007/2010 Integration	Yes	No
Centralized operation.	Yes.	No.
Queue Announcements	Yes.	Yes.
Auto Attendant	Yes.	Yes.
Call Recording	Yes.	No.
Intuity Emulation	Yes.	No.

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1. Up to 40 in a single-site environment, additional ports can be added in a SCN environment by Distributed VoiceMail Pro.

2. The first two ports are included in the basic IP Office system price. Additional 4 ports(sold in pieces of 2) can be purchased and licensed separately.

IP Office Essential Edition - Embedded VoiceMail

In environments like retail or home office, where space, noise or cost considerations rule out using a PC for voicemail, Embedded Voicemail will be the favored option for an entry-level voicemail service. As built-in functionality of the IP500 and IP500 V2 hardware it requires no separate server. The Embedded Voicemail is enabled by purchasing the IP Office Essential Edition.

Key features of Embedded Voicemail include:

- Up to 6 ports of voicemail on IP500 V2
- IP500 V2: The first 2 ports are included in the basic system. Up to four additional ports can be added simply by purchasing two 2-port add-on license keys
- IP500: Maximum 4 ports of voicemail
- With an increasing number of ports the message storage capacity is increased simultaneously:
  - 2 ports: Up to 15 hours
  - 4 ports: Up to 20 hours
  - 6 ports: Up to 25 hours
- Configurable recording time: Default value 2 minutes, maximum value 3 minutes
- Multiple languages stored on the system
- Help menus via *4. Greetings and mailbox navigation.
- Send an email containing either notification or the complete voicemail message
- Voicemail breakout/personal auto-attendant: Up to 3 breakout numbers can be set up. When callers are directed to your mailbox, they can either leave a message or choose to be transferred to one of three numbers (e.g. Operator, mobile/cell phone, colleague, etc).
- Configurable system-wide short code for voicemail access (e.g. *17)
- 40 configurable Auto Attendants with 3 time profile base greetings per Auto Attendant
- Up to 12 menu items per Auto Attendant with automatic time-out to fallback number.
- Auto attendant supports Dial by Name and direct Dial by Number (dial extension number without dialing access digit first)
- Access and control of voicemail via the digital or IP telephone display (Visual Voice). This feature is supported on the 1408, 1416, 1608, 1616, 2410, 2420, 4610, 4620, 4621, 4625, 5410, 5420, 5610, 5620, 5621 and 9600 series telephones.
- Reply to a message to either an internal or external number (if Caller ID available)
- Support for hunt group announcements
- Fax option for rerouting fax calls via the auto-attendant menu
- Support for Fast Forward (#), Rewind (*), Skip Message (9) and Call Sender (**) when listening to messages
- No license required

IP Office Preferred Edition - Voicemail Pro

IP Office VoiceMail Pro is enabled by the IP Office Preferred Edition and is the most advanced messaging and call flow application for IP Office systems. It can handle up to 40 simultaneous calls depending on license and system settings. Each user has the option of turning their voicemail on or off. When on, the system automatically answers their telephone when they are not available to take a call, plays a personal greeting, and records a message.

When a message has been left, the user will see a message-waiting lamp lit on their telephone and can press a retrieval button to collect their messages.

VoiceMail Pro can also ring the user to deliver any new messages. Voicemail messages are time and date stamped and the caller's number recorded. VoiceMail Pro can be configured to delete read messages automatically, unless the user chooses to save the message permanently.

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Voicemails can be collected remotely by dialing into the VoiceMail Pro server. If the number the user is dialing from is recognized (home number or mobile/cell phone for example), the user will listen to their voicemail straight away. If the source number is not recognized, the user will be prompted for a mailbox number and a PIN code for that mailbox, before they can listen to their voicemail. Users have the ability to set and change their own PIN codes.

When a voicemail needs to be forwarded to other users, VoiceMail Pro provides many options:

- Voicemails can be forwarded to another mailbox, or group of mailboxes
- Recipients can add their comments to the voicemail before forwarding to another mailbox or mailboxes.
- Voicemails can be forwarded as email WAV attachments.

All options are available in a choice of languages; both spoken voice prompts and graphical programming interfaces and have the choice of IP Office TUI and INTUITY emulation TUI.

Call Flows with VoiceMail Pro for Intelligent Call Handling

At the heart of VoiceMail Pro is the ability to construct call flows from a series of different building blocks. These building blocks allow automation over tasks like answer a call, listen for tone-dialed digits, make a call etc. VoiceMail Pro call flows allow far more than just guiding a user to the group or extension they require. Call flows allow VoiceMail Pro to dial back users as soon as a voicemail message is left for them, it provides remote access to phone forwarding settings should a user wish to change their Forwarding or Follow Me number from an external telephone. VoiceMail Pro provides message handling for individuals or groups, audio information to callers so assisting the operator during periods of heavy call activity and links to business applications through services such as Text-to-Speech. VoiceMail Pro provides a full telephony applications environment where call flows can be set up and interact in real time with business workflow – callers can interact via menus and data entry and VoiceMail Pro applications can speak back results. For example, users can listen to their email messages through the telephone.

Email Integration

Integration with email systems is provided by the VoiceMail Pro Unified Messaging Service (UMS) which is delivered to eligible users as part of the Office Worker, Teleworker or Power User license. It enables VoiceMail Pro to interact with email systems to provide a synchronization of voicemails and their status (new/unread, read, deleted, saved) between all user devices like desk telephones, mobile cell telephones, the UMS web interface, Avaya one-X™ Portal for IP Office, the email client and other devices that are synchronized with the user's email account. This will work for email clients that use the IMAP4 protocol (like MS Outlook, Lotus Notes, Mozilla Thunderbird, and many others).

VoiceMail Pro and Exchange Server 2007 or 2010

Advanced collaboration between VoiceMail Pro and Microsoft Exchange Server 2007/2010 offers voicemail storage into the Exchange message store. This is the single point of storage for all email and voicemail messages and therefore the single source for all message status information. When a voicemail should be collected using a desk phone connected to the IP Office, VoiceMail Pro will retrieve it directly from the Exchange message store. VoiceMail Pro sends the voicemails not just as emails with a .WAV attachment but as a message formatted as 'voicemail' so that Exchange can handle them differently from emails.

If Microsoft Exchange Server 2007/2010 working with a mobility solution server (e.g., a Blackberry Enterprise Server) are used to push emails to mobile devices (Blackberry or similar), VoiceMail Pro will also integrate and deliver voicemails via the Exchange Server to the mobility solution. The mobile device will then be able to present voicemails on the visual voicemail interface (instead of presenting them as emails with an attachment).

Voicemail in a Small Community Network (SCN)

A single PC based VoiceMail Pro server can provide voicemail services to multiple IP Office systems in a Small Community Network over the LAN, WAN or a Frame Relay network. This is referred to as 'Centralized Voicemail' and can reduce costs, while facilitating communication between IP Office sites. For resilience, should the main site fail, the Centralized Voicemail will automatically reconnect to an alternative IP Office site.

For Voicemail connections from remote IP Office users in a SCN no multi-site channels are needed.

VoiceMail Pro - More than just Voicemail

VoiceMail Pro offers much more than just a pure voicemail system. Other helpful, convenient, cost and time saving options are:

- Whisper Announce that prompts callers for information (usually their name) which is recorded and passed on to the user's extension on answer, allowing them to choose to accept the call or not. This is particularly useful on "CLI/ANI withheld" numbers - usually calls from telesales companies where somebody is trying to sell you something. VoiceMail Pro will not intrude onto busy extensions.
- Assisted Transfer allows transfer of a call to a destination, but allows the call to return to VoiceMail Pro automatically for other options should the called party be engaged, or not answer within a pre-determined time.

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- Conditional routing of calls. Conditions are constructed from a set of basic elements. These elements can be combined within a single condition to create complex rules. For example, the Week Planner can be used to define the company's standard working hours, and then combined with the calendar to define exception days such as public holidays / vacation.
- Call modules. Modules allow you to create sequences of actions that you want to share between a number of different call routing scenarios – like a "macro" in PC applications. These modules can be used to create a library of vertical voicemail applications or just easy dissemination to other IP Office voicemail sites, thanks to its import and export functionality.
- Activation of the external relays on the IP Office system. For example, remotely checking the status of the office meeting and then turning it on from your mobile/cell phone on your drive in to work.

Key features of VoiceMail Pro include:

- Personal voice mailbox for users and hunt groups.
- Personal Numbering (follow me).
- Extended personal greetings to customize the information presented to a caller based upon the availability of a user.
- Unified Messaging (UMS) offers voice mail - email synchronization between the VoiceMail Pro server and the email client.
- UMS Web Access allows access to voice mails via a web interface from an internet browser.
- UMS integration of VoiceMail Pro with Microsoft Exchange 2007 Server for full message synchronization.
- Enables mobile messaging integration (e.g., Blackberry) when used with Exchange 2007 server integration.
- Forwarding of voicemail messages to email systems via SMTP.
- VoiceMail Pro client, a graphical user interface for programming and configuring applications both locally and remotely.
- Data base access via Interactive Voice Response (IVR) for individual business requirements.
- Audiotex and Auto Attendant services (including dial by name).
- Sophisticated queue announcement facilities.
- Access and control of voicemail via the digital or IP phone display (Visual Voice).
- 22 supported prompt languages: Chinese (Mandarin), Danish, Dutch, English (UK), English (US), Finnish, French (France), French (Canadian), German, Greek, Hungarian, Japanese, Italian, Korean, Norwegian, Polish, Portuguese (European), Portuguese (Brazilian), Russian, Spanish (Castilian), Spanish (Latin American), Swedish.
- Conditions (e.g. test if 'out of hours').
- Broadcast group messages.
- Automatic and on demand call recording with an option for ContactStore search and replay of saved messages.
- Tampering Detection / Verified Call Recording
- Voice forms/questionnaire mailboxes (Campaign Manager).
- Personal distribution lists.
- Tag information retrieved from a database to a call and delivers it with the call to an agent.
- Visual Basic (VB) Script support to allow the configuration of the voicemail system through VB scripts rather than VoiceMail Pro call flows.
- Text-to-Speech facilities to allow emails to be read out over the telephone and/or for database information to be read to a caller in 14 languages.
- Housekeeping facilities for the management of messages.
- Automatic detection and routing of fax calls within Auto Attendants and within a subscriber's voicemail box.
- Support for a range of the INTUITY telephone user interface features in INTUITY emulation mode.
- Recording of system prompts through the telephone handset or using multimedia facilities on a PC.
- Speaking clock.
- Support for TTY hearing impaired text phone.

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- Centralized VoiceMail within a multi-site IP Office environment.
- Networked Messaging with other Avaya voicemail systems.
- Capacity of up to 40 ports for a single-site system.
- Voicemail channels between VoiceMail Pro and the IP Office can be reserved for business critical functions or left unreserved for any function.
- Improved voice recording, including recording of calls made over IP telephones (calls using direct media has to be routed through IP Office); automatic call recording triggered by incoming call routes; pausing recording when call is parked or placed on hold.
- User start points in VoiceMail Pro include queued options.

Further details on some of the VoiceMail Pro functionality listed above are described later in this section.

IP Office Preferred Edition - VoiceMail Pro Resilience and Backup

Customer service with uninterrupted access to the business is essential – for customers who want to place orders, for partners who want to negotiate an important contract and for employees who need access to their important messages. A failure can be very expensive for a business and there are a number of reasons why a failure can happen: power outage, lost network connection, cable cuts, hardware failure, and so on.

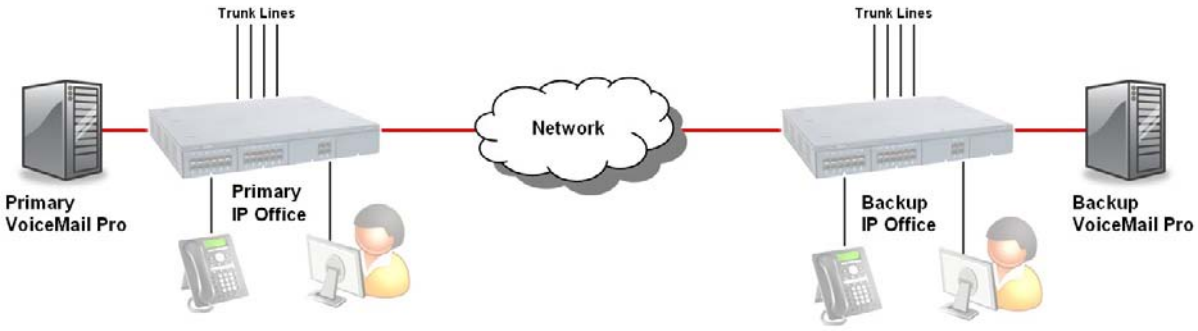
But loss of money and reputation does not necessarily have to follow such a failure. The IP Office Resilience and Backup functionalities are supported by VoiceMail Pro. A VoiceMail Pro server, connected to a Primary IP Office, can automatically re-connect to a Backup IP Office if the connection between the two IP Office systems gets lost. This covers the failure of an IP Office or the connection between them. A Preferred Edition license is required on the backup IP Office system for this scenario.

Introduced with Release 6, a Backup VoiceMail Pro server offers additional comprehensive functionalities to keep the business communication up and running. A Backup VoiceMail Pro server can be added to an IP Office / VoiceMail Pro system to take over the voicemail functions in the case of a failure at the primary VoiceMail Pro system.

The Backup VoiceMail Pro server can act in different scenarios:

- One IP Office, one Primary VoiceMail Pro server, and one Backup VoiceMail Pro server, where the backup is either at the same location as the IP Office / VoiceMail Pro or at an external site
- A Primary IP Office with a Primary VoiceMail Pro server and a Backup IP Office with a Backup VoiceMail Pro server at a second location
- A Multi Site Network with a Centralized VoiceMail Pro server connected to an IP Office at one site and a networked IP Office with a Backup VoiceMail Pro server. There might also be additional IP Office and Distributed VoiceMail systems within that Multi Site Network.

The diagram illustrates the second scenario:



The Primary VoiceMail Pro will establish a link to the Backup VoiceMail Pro and keep the greetings and configuration setting changes synchronized with the Primary VoiceMail Pro. Messages and message states are not synchronized between the Primary and the Backup VoiceMail Pro to avoid network overload.

Ideally the Backup VoiceMail Pro server is a mirror system of the Primary VoiceMail Pro server with all its capabilities, like the number of ports, licensed features, and hardware. All required functions have to be licensed through IP Office Preferred Edition and supported by adequate hardware (e.g. to ensure server performance and sufficient IP Office trunk lines). The 2 hour run-time of a VoiceMail Pro server without licenses does not apply to a backup scenario.

The Backup VoiceMail Pro server can have limited functionalities (e.g. fewer number of ports) if it is acceptable to have limited capabilities for a limited period until the re-connection of the main server.

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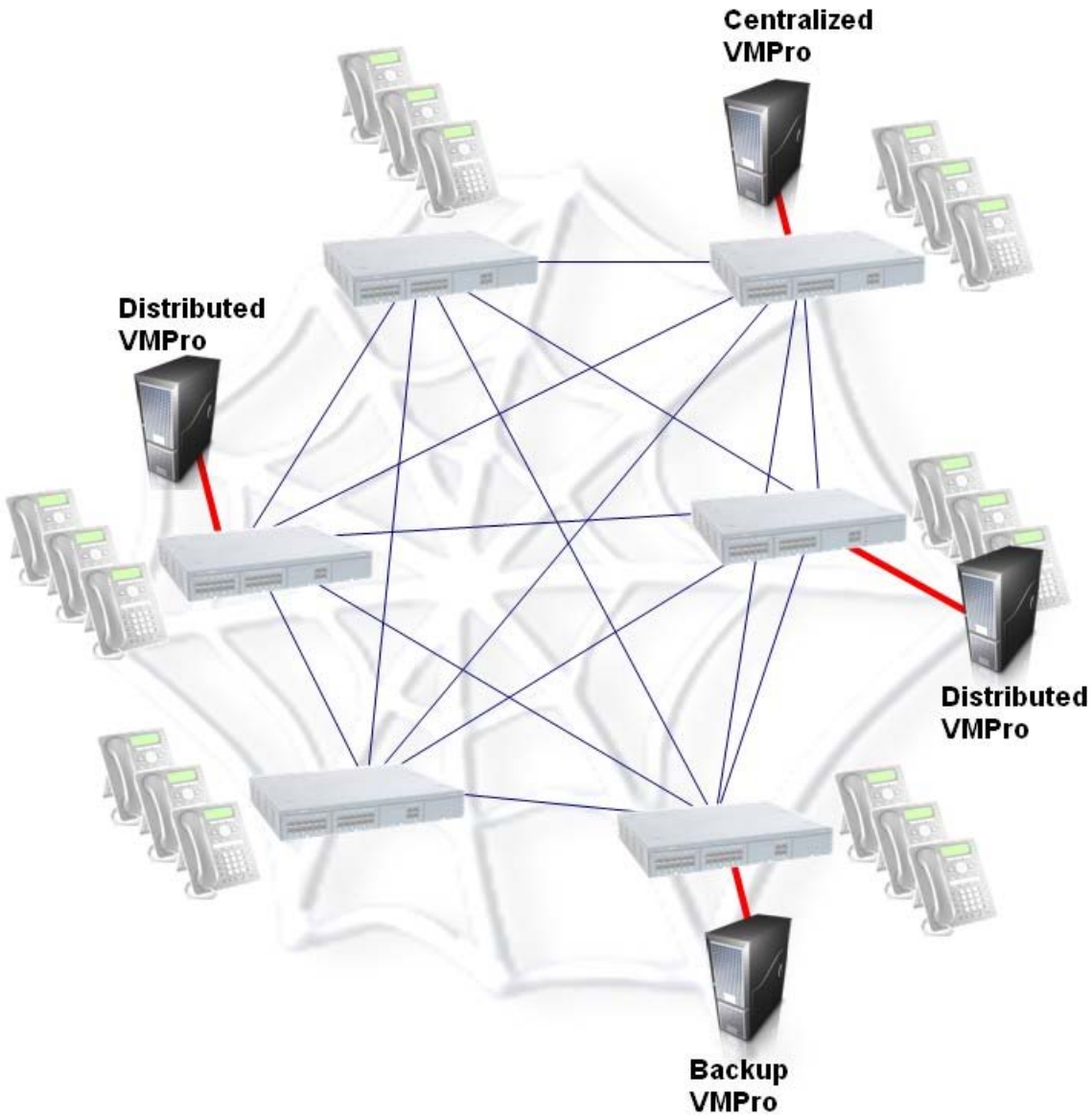
Distributed Messaging

One Centralized VoiceMail Pro server is supported in a SCN of multiple IP Office systems with IP Office / VoiceMail Pro release 5 and earlier. IP Office R6 allows multiple Distributed VoiceMail Pro servers to be placed at selected or all of the IP Office nodes.

Into one Multi Site Network there can be one Centralized VoiceMail Pro, one Backup VoiceMail Pro (as a backup for the Centralized VoiceMail Pro), and up to 30 Distributed VoiceMail Pro (theoretical value). The Centralized VoiceMail Pro server will keep its role as the main voicemail server. It provides the media store for voicemails left on the system, as well as the media services required on the Multi Site Network to IP Offices not supported by a local VoiceMail Pro server.



The retrieval of voicemails will require access to the Centralized VoiceMail Pro, while other voice resources will be available locally, providing local capabilities for queuing announcements, call recordings, auto attendants and leaving of voicemails.

On failure of a Distributed VoiceMail Pro, the Centralized VoiceMail Pro will be used for all functions as if no Distributed VoiceMail Pro servers were present.

Centralized VoiceMail Pro, Distributed VoiceMail, or Backup VoiceMail require the same Preferred Edition license but will be configured differently in IP Office Manager according to the required function.

Interaction of Voicemail with Email Systems

As standard, VoiceMail Pro allows for a simple voicemail alert where the entire voicemail is forwarded (copied) as a .WAV attachment to any MAPI or SMTP compliant email application (Microsoft Outlook, Exchange, Lotus Notes, etc.). Forwarding allows emails and voicemail messages to be unified and collected from a single source. This simple alert option that forwards only the caller's number in the subject of the email is ideal for use with commercial Short Message Systems (SMS). This information can be forwarded to the display of a mobile/cell

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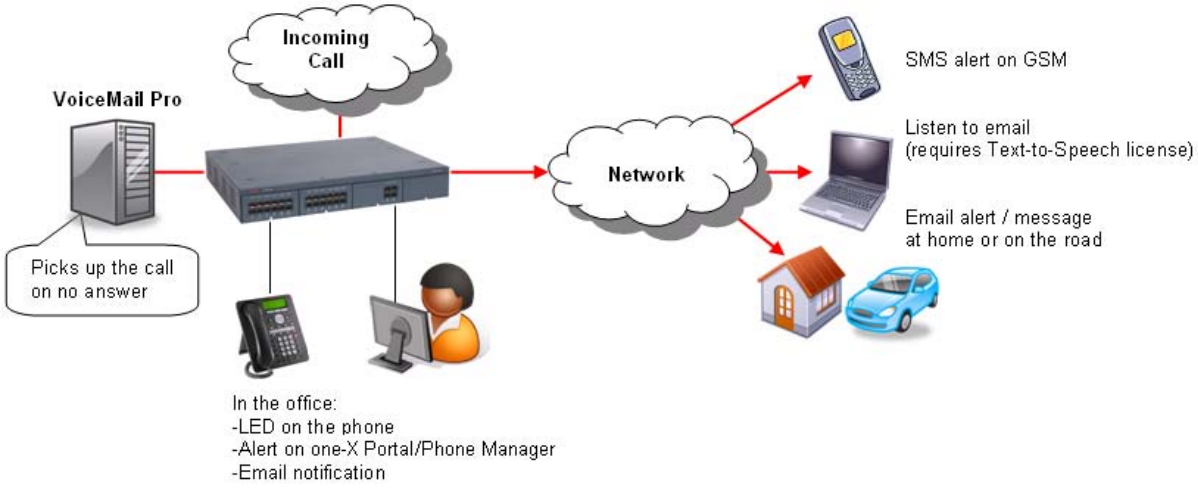
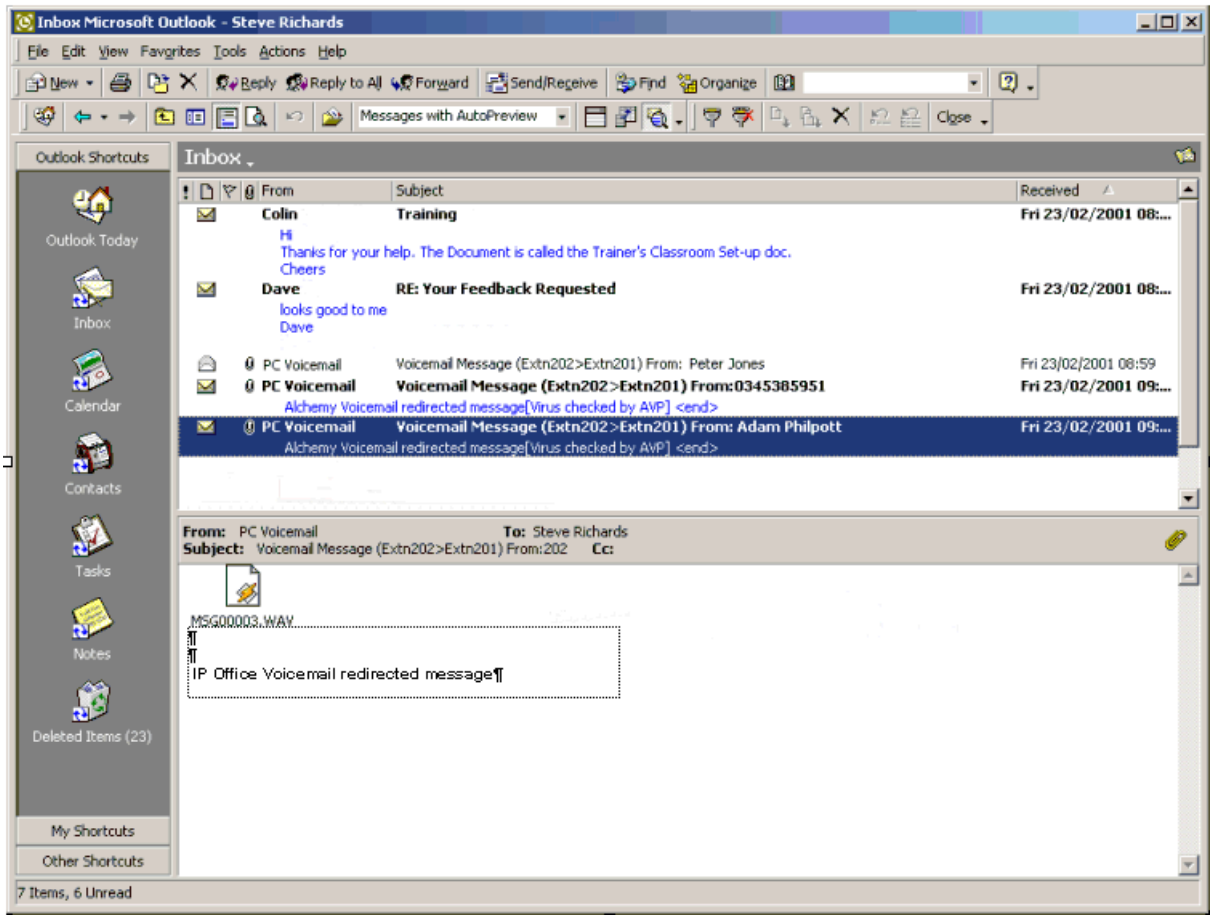
phone when the user is away from the desk. The email notification, forwarding and copying, can be done for all voice messages and can be activated remotely. This is beneficial if you are working from home and have an email connection available.


Incoming Call
VoiceMail Pro
Picks up the call on no answer
Network
SMS alert on GSM
Listen to email (requires Text-to-Speech license)
Email alert / message at home or on the road
In the office:
-LED on the phone
-Alert on one-X Portal/Phone Manager
-Email notification

Forwarding voicemail to email is particularly useful for group voicemail boxes as it allows a single voicemail message to be copied to the email of every member in that group.



Presentation of Voicemail to Email

Unified Messaging Service (UMS)

Unified Messaging Service (UMS) provides the integration of VoiceMail Pro with email systems. A very simple installation and configuration process enables users to manage their voicemail messages by using the device of their choice including email clients or mobile devices. UMS is licensed on a per-user basis as part of the IP Office Power User, Teleworker and Office Worker.

Depending on the existing infrastructure UMS can be integrated with IMAP-based email solutions as well as with Microsoft Exchange Server 2007/2010 environments.

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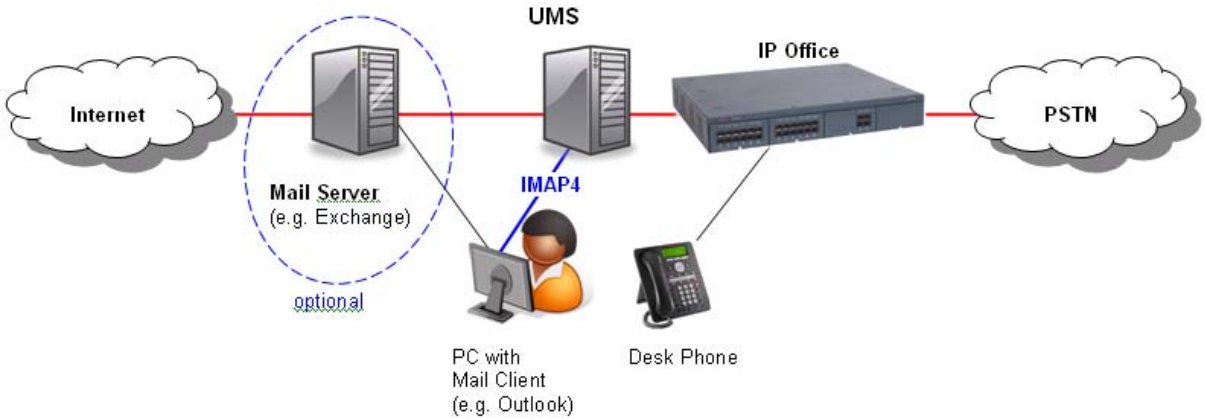
Voicemail Synchronization via IMAP

Email applications such as Outlook which support IMAP can connect to an IMAP server integrated with the VoiceMail Pro server.

VoiceMail Pro will deliver new voicemail messages by using the voicemail to email capability.



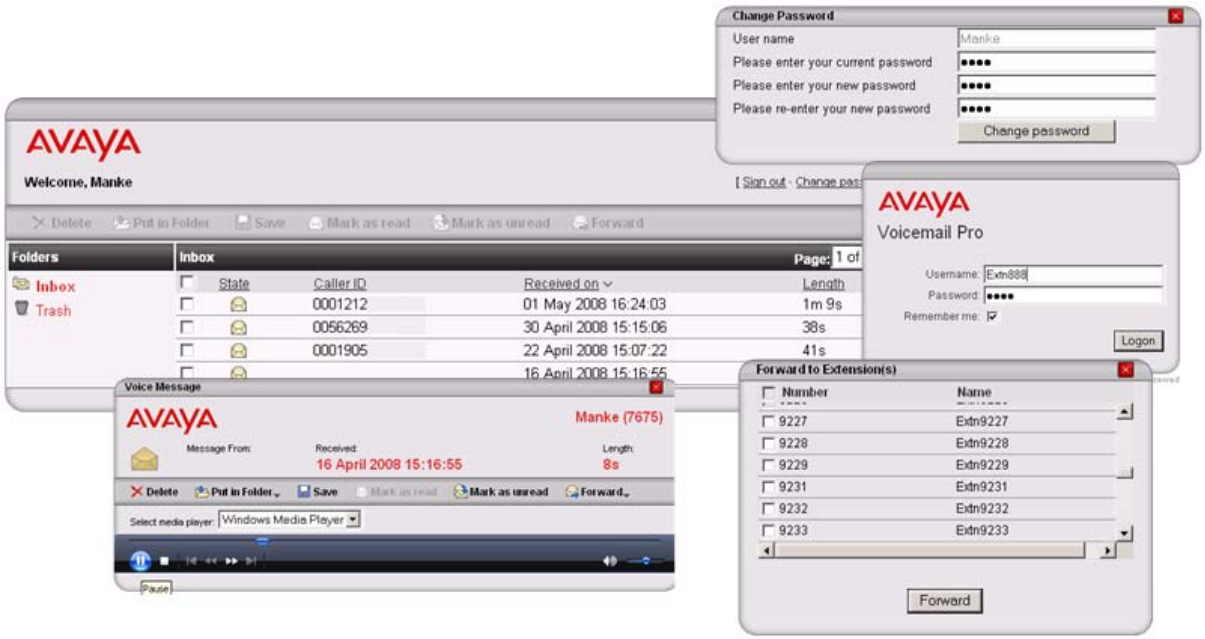
New voicemails arrive as emails with a .WAV attachment and the user can handle them like any other email. If a voicemail has been listened to, its status will change to "read", independent of the device that has been used to access the voicemail. If the user deletes it, it will be deleted everywhere.

Users that have UMS access enabled just need to add an additional email account to their email client to be able to use the unified messaging features. No additional client software installation is required.

The instructions how to add this account and configure the web address for the VoiceMail Pro Web Access can be provided by an administrator by email without the need for system administration or external support.

VoiceMail Pro UMS Web Access

The second interface UMS offers to the users is a web interface allowing voicemail access via a web browser, such as Internet Explorer or Mozilla Firefox, with the ability to listen to targeted voicemails by using either the PC's multimedia equipment or the desk phone.



UMS Web Access requires the VoiceMail Pro installation to be on a server with IIS pre-installed. In addition, the VoiceMail Pro UMS Web Access option must be selected during installation.

Once installed, the system will require the following additional configuration:

- IP Office Power User, Teleworker and Office Worker licenses (for the number of users that will be configured to use UMS).
- Users selected for UMS support must be enabled by the system administrator in IP Office Manager.
- Each user must have a voicemail PIN code (a blank PIN is not acceptable).

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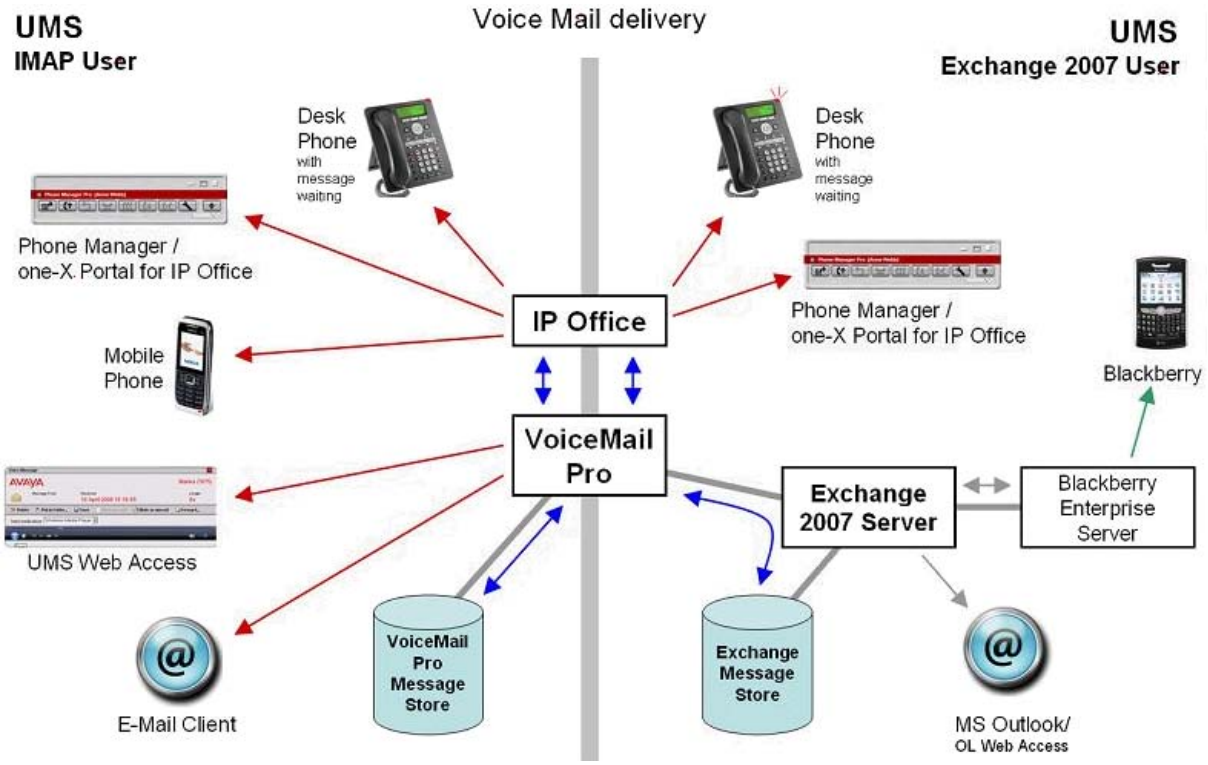
Integration with Microsoft Exchange Server 2007 or 2010

As part of UMS, VoiceMail Pro can integrate with the Exchange Server 2007/2010 from Microsoft. Each user with UMS enabled can be configured to use either the IMAP capability (see above) or the Exchange integration. If the Exchange option is selected all voicemails of the user will be forwarded and stored into the Exchange 2007/2010 message store instead of the VoiceMail Pro message store. VoiceMail Pro is able to deliver new voicemails to Exchange formatted as 'voicemail' which enables Exchange to handle them differently from standard emails. If used in combination with a mobility solution voicemails can be managed using the visual voicemail user interface of the mobile device (e.g. Blackberry).

The following diagram illustrates the two possible ways of integrating with either a standard IMAP-based email system or with an Exchange Server 2007/2010 email system.



Both scenarios can exist in parallel on the same system but only one option can be configured for one user.

A voicemail stored in the Exchange message store can be retrieved by VoiceMail Pro and made accessible for all communication devices (as shown in the diagram above).

Earlier versions of Microsoft Exchange Server are supported by using the IMAP integration described above.

Fax Messages

Integration with fax software, integration with fax to the desktop or client fax applications can be realized through the use of fax servers. This allows an email client (for example Microsoft Outlook) to be utilized as an easily affordable unified messaging solution. The many benefits of unified messaging include security (as faxes are sent to the users PC rather than on paper for everyone to see), ease-of-use and efficiency in terms of storage and retrieval of messages and the great gains that can be made in overall workforce efficiency and productivity.

To enhance the support of third party fax solutions, VoiceMail Pro supports the automatic detection of incoming fax calls. Traditionally a dedicated telephone number is provided for all incoming fax calls. In addition to, or as an alternative to, the VoiceMail Pro 'Menu' action or a subscriber's voicemail box (INTUITY mode) can automatically detect any incoming fax calls and then direct the call to a predefined location. The benefit to a business or user is that only one number is required for either voice or fax calls.

The VoiceMail Pro can store the default fax location for the automatic routing of fax calls. Alternatively, with fax tone detection at the voicemail box, each voicemail box can have a fax location number. If a voicemail box owner has set his or her own fax number, then that number is used instead of the default fax location.

Voicemail box subscribers can set their own fax number through their mailbox menus.

Most fax solutions can be used in conjunction with IP Office; however the following products have been tested and verified to operate in the above scenarios:

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- Equisys - Zetafax
  Zetafax for Networks provides versatile network fax software solutions for small businesses, corporate offices and distributed enterprise businesses. It enables employees to send and receive faxes at their desktop, without the need to print fax communications, take them to a fax machine and send them manually. Zetafax can be seamlessly integrated into market leading email systems like Exchange allowing users to send and receive faxes directly from their Outlook client. In addition Zetafax can be integrated with other existing applications, such as accounting or CRM systems, for fast, automated faxing from the desktop or back office. Further product information available from www.equisys.com.
- Open Text Fax Server (formerly Captaris – RightFax)
  RightFax offers a broad, scalable product line that integrates with email, desktop, CRM, ERP, document management, imaging, archival, call center, copier/scanner systems, as well as host, legacy and mainframe applications–virtually all business applications. Further product information available from www.captaris.com.
- Fenestrae – Faxination
  Fenestrae Faxination Server for Microsoft Exchange integrates fax into email technology. Create faxes on your desktop and deliver them to your chosen fax machine at the click of a mouse. Further product information available from www.fenestrae.com.
- GFI – GFI FaxMaker
  GFI FaxMaker for Exchange/SMTP allows users to send and receive faxes and SMS/text messages directly from their email client. It integrates with Active Directory and therefore does not require the administration of a separate fax user database. GFI FAXmaker integrates via the SMTP/POP3 protocol with Lotus Notes and any SMTP/POP3 server. Further product information available from www.gfi.com.
- Open Text Fax Appliance (formerly Castelle FaxPress)
  Faxes routed to a user's mailbox by this fax server will be recognized by VoiceMail Pro as faxes, and will be supported by VoiceMail Pro Fax features. More information is available from www.castelle.com.

Text-to-Speech Email Reading (Microsoft Exchange only)

In addition to providing a unified mailbox for voicemail messages, emails and fax message, VoiceMail Pro can also provide the ability to retrieve email messages through the telephone.

When operating in INTUITY mode and with Power Users and/or Mobile Workers licensed these users will be presented with a list of both voicemail messages and email messages. The emails can then be read out over the telephone. The benefit to the user is that their messages are now accessible while in and out of the office through any telephone. The user's email account needs to be configured in IP Office manager to enable TTS functionality.

When accessing messages through the telephone all new voicemail messages will be presented to the mailbox owner before any new email messages. When accessing an email message the system refers to the message as *"New message with text"*.

Emails will be read based on the user language setting in IP Office, depending on the TTS engine capability. With no additional TTS engine installed VoiceMail Pro TTS will use the TTS engine installed with the Microsoft Windows operating system. If using Avaya TTS then 22 languages are supported. If using 3rd party TTS please refer to the vendor documentation for supported languages. Avaya TTS is an optional license.

Where the user has email reading in their voicemail box, they will be able to record a voice reply to the email, and send it as a .WAV attachment to a reply email to the person who sent the email.

A maximum of eight users can use TTS functionality simultaneously. If more than eight users are licensed and configured for TTS, the rule 'first come – first serve' will apply. In an SCN with a Centralized VoiceMail Pro the limit of eight concurrent TTS users for email reading will apply for the whole SCN because only the Centralized VoiceMail Pro delivers TTS. Therefore it is necessary to license at least one Power User or Mobile Worker at the central site (better will be four to eight users to provide sufficient resources for all users in the SCN).

Networked Messaging

Where organizations are operating a number of voicemail systems across different sites it is important to be able to provide integrated operation between voicemail systems so that messages can be passed between systems and delivered to a user's mailbox seamlessly. This is achieved by IP Office VoiceMail Pro being licensed to support Networked Messaging.

The Networked Messaging solution defines a common set of features to allow inter-working between Avaya voicemail systems. In INTUITY mode, while listening to or having listened to a message, the user can select the option to forward the message to another mailbox, the mailbox entered can be any mailbox number on the local system or any mailbox on a remote Avaya system.

The IP Office Networked Messaging facility will allow configuration of up to 2000 remote mailboxes on each VoiceMail Pro server and will operate with other IP Office systems supporting this feature, as well as with Avaya Enterprise solutions.

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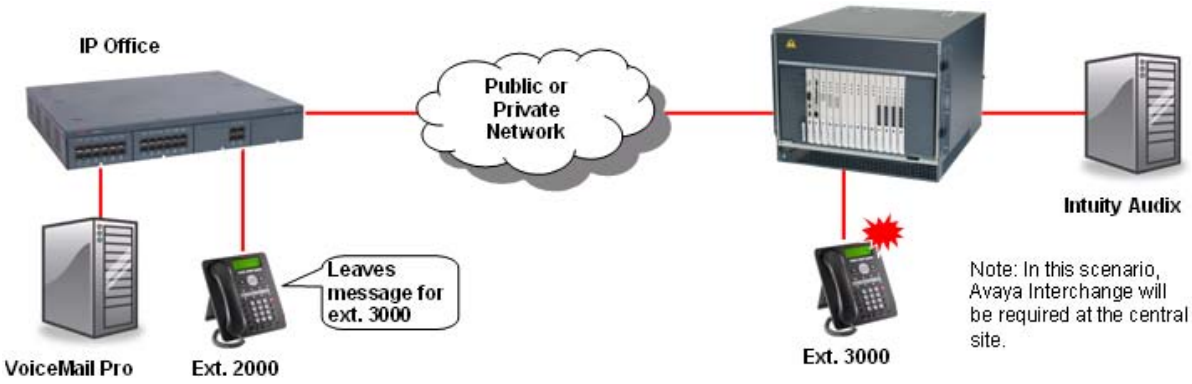
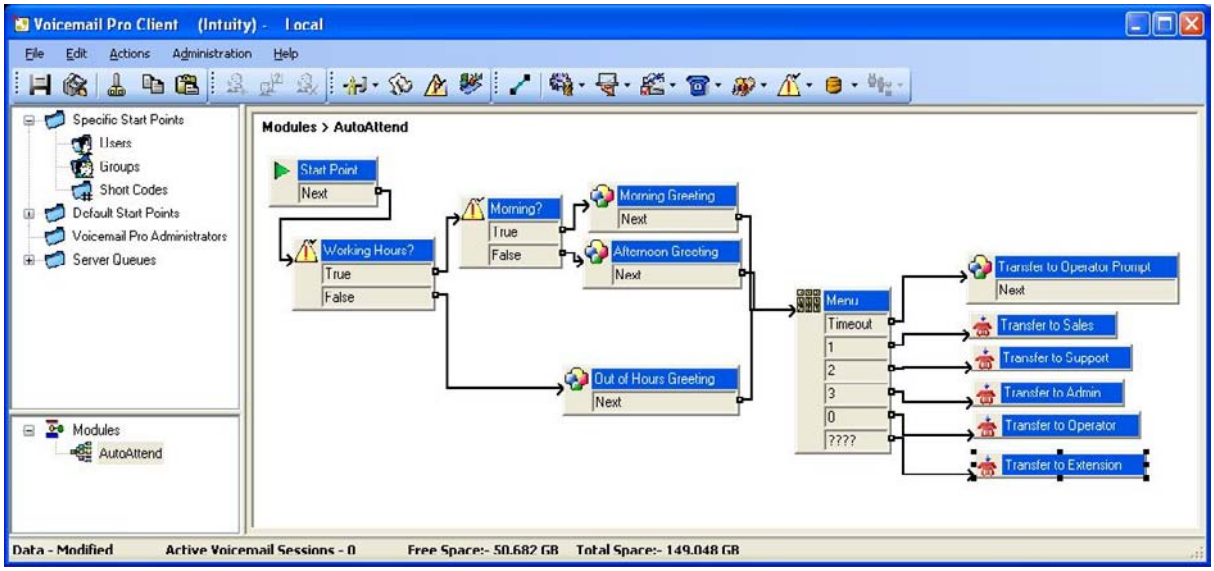

IP Office
Public or Private Network
Intuity Audix
Leaves message for ext. 3000
VoiceMail Pro
Ext. 2000
Ext. 3000
Note: In this scenario, Avaya Interchange will be required at the central site.

Auto Attendant

VoiceMail Pro provides an easy-to-use, multi-level configuration tool (the VoiceMail Pro client) which allows network managers and system administrators to construct an interactive menu system, based upon DTMF telephone key entry. This allows an Auto-Attendant system to be built and configured to suit business needs, be that on its own or as a back-up for the regular operator when call volumes are high. VoiceMail Pro offers the caller the ability to dial the name of a person via the phone keypad (like "Text" messaging on mobile cell telephones). In response the Auto-Attendant offers the caller a best match name or if there is more than one, a selection list is offered and the caller can select which one they want to call.

As an example, VoiceMail Pro can be used to build an Auto-Attendant that prompts callers to "enter 1 for sales, 2 for support, 3 for admin, or 0 for the operator" allowing them to be transferred to the appropriate department without operator intervention. Alternatively, a list of personnel and their extension numbers could be listed, allowing the caller to directly access the person they want. For larger companies it could be department name listed first, followed by the list of employee extensions within the department.

The latter two examples are ideal where company telephone operation has changed from a central operator to direct dialing (DDI/DID), allowing callers to "learn" the required extension number from the prompting of VoiceMail Pro, and then in future dial the extension number, or other pre-defined variables, directly. Auto-Attendant operation is also ideal where multiple languages are required, for example *"Dial 1 for English, 2 for German, 3 for French, ..."*.



Auto-Attendant created using VoiceMail Pro Manager

Personal Numbering

Contact-ability is all-important in winning and maintaining business. VoiceMail Pro offers users the ability to remotely turn their voicemail on or off, set their voicemail email forwarding, edit their call forwarding and follow me numbers. Together these actions provide a comprehensive Personal Numbering service for the user who needs to remain in contact regardless of their physical location.

Users with Mobile Twinning configured are able to remotely activate their twinning capabilities through VoiceMail Pro call flow.

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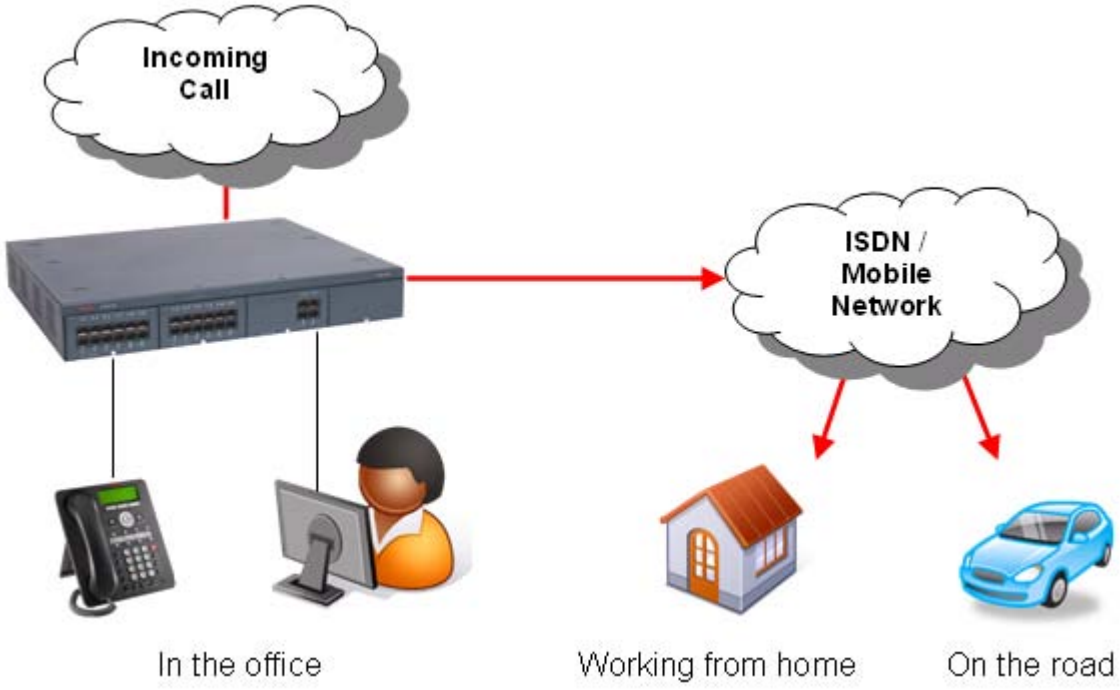

Diagram illustrating Personal Numbering

Accessing Database Information within Call Flows (IVR)

VoiceMail Pro provides the ability to construct powerful interactive systems based upon DTMF telephone key entry. This is achieved by using the flexibility provided from the built-in call flow actions. As a caller passes through any part of a defined call flow the system is capable of interacting with most third party databases using the standards based ADO interface (ActiveX Data Objects). The system is capable of retrieving information from a database and writing information into databases. The result of this is that powerful Interactive Voice Response systems (IVR) can be delivered to specifically meet the requirements of the business and the customer experience that is required.

Example interactive systems that can be built as a result of these facilities include: Information bulletin boards, order taking and order processing systems, front end systems to help desks/support desks, contact centers, secure access to information through PIN checking, survey systems, remote time sheet management, etc.

The ability to interact with database information is enabled through the purchase of the IP Office Advanced Edition. The VoiceMail Pro Client supports four Database Action Icons, which can be implemented into the VoiceMail Pro call flows. The entry of the IP Office Advanced Edition license will enable the VoiceMail Pro server to use these actions in call flows.

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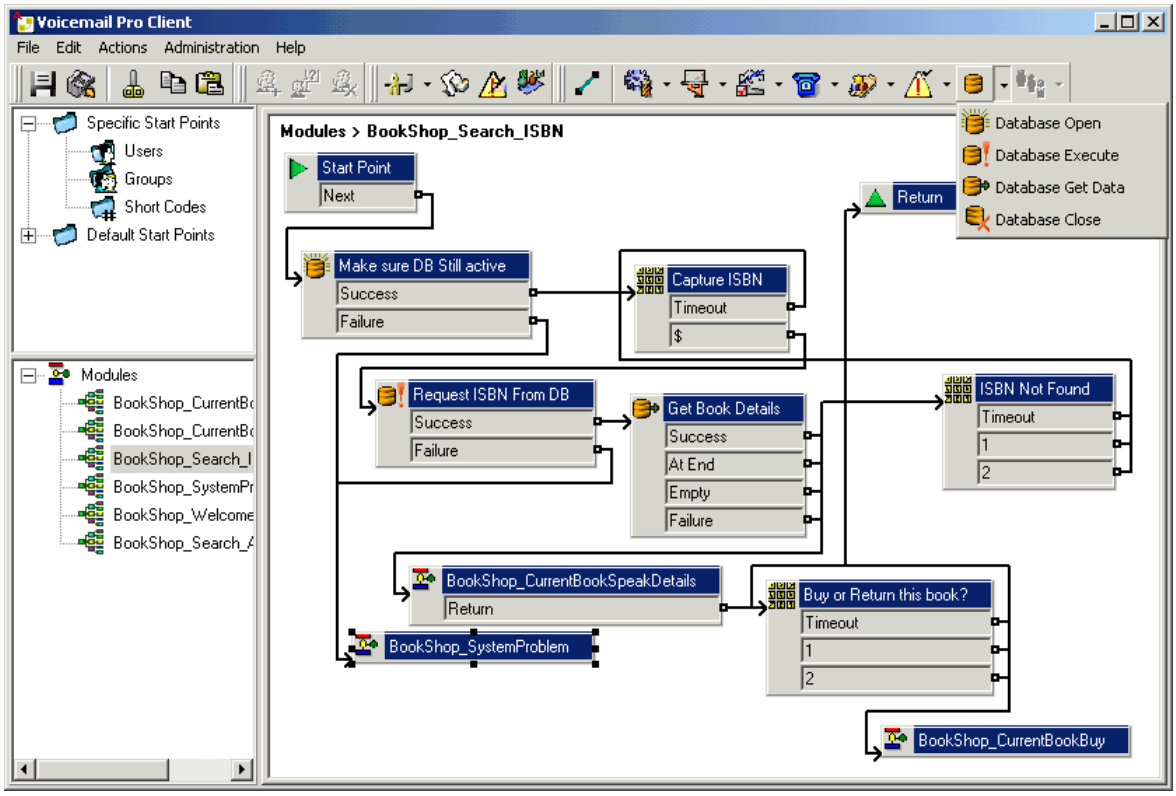

Example Call Flow Utilizing Database Actions

The database actions that are provided through the VoiceMail Pro Client are:

- Database Open – Opens a link to the required database. Multiple databases can be accessed during a call but only one database can be opened at one time.
- Database Execute – Provides the ability to enter a query on the opened database. The query can 'Select' data from the open database or can 'Insert' data into the database.
- Database Get Data – Provides access to the data that has been retrieved from a database through the Database Execute action. The user can retrieve the next item, previous item, first item in the list or the last item in the list.
- Database Close – This action will close the current database. If the database is open when a call terminates then the database will be automatically closed.

Interaction with the opened database is done through Structured Query Language scripts (SQL). An administrator can enter SQL script directly into the specific section of the Database Execute action. For administrators that are not familiar with SQL scripts, a script can be created automatically through the use of a SQL Query Builder Wizard.

Using Text To Speech (TTS) Facilities within a Call Flow

Text-to-Speech (TTS) can be used to further enhance IP Office IVR and call flow capabilities; TTS facilities can enhance the callers experience by allowing the system to read back to them any information that has been extracted from a database. For example, in a book shop, the caller dials into the system and is asked for an ISBN number of the book they require. The caller enters the ISBN through the telephone keypad and the system locates the title of the book from the database. As well as finding the title, the system could also look up the author of the book and whether there were any books in stock. By using TTS, the system could now respond to the call:

"The book, Lord Of The Rings, costing $6.99, written by J R R Tolkien is in stock".

From VoiceMail Pro Release 6 onwards it is also possible to use Text-to-Speech within call flows or auto attendants for announcements. A call flow / auto attendant can be enabled to read text that has been created by call flow actions and offers much more flexibility because there is no longer the need for pre-recorded and static announcements. If there is a change to an announcement the call flow action will create the new announcement and will play it to the caller.

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The IP Office Advanced Edition contains 8-port 3rd party Text-to-Speech capability and can be used either with a TTS engine that has to be purchased separately or - when no 3rd party engine is installed - with the Microsoft TTS engine which is part of the Windows operating system.

An alternative is the Avaya TTS license which adds a TTS engine available in 22 different languages:

- Chinese (Mandarin)
- Chinese (Cantonese)
- Danish
- Dutch
- English (UK)
- English (US)
- Finnish
- French (Standard)
- German
- Greek
- Japanese
- Italian
- Korean
- Norwegian
- Polish
- Portuguese
- Portuguese (Brazilian)
- Russian
- Spanish
- Spanish (Latin)
- Swedish

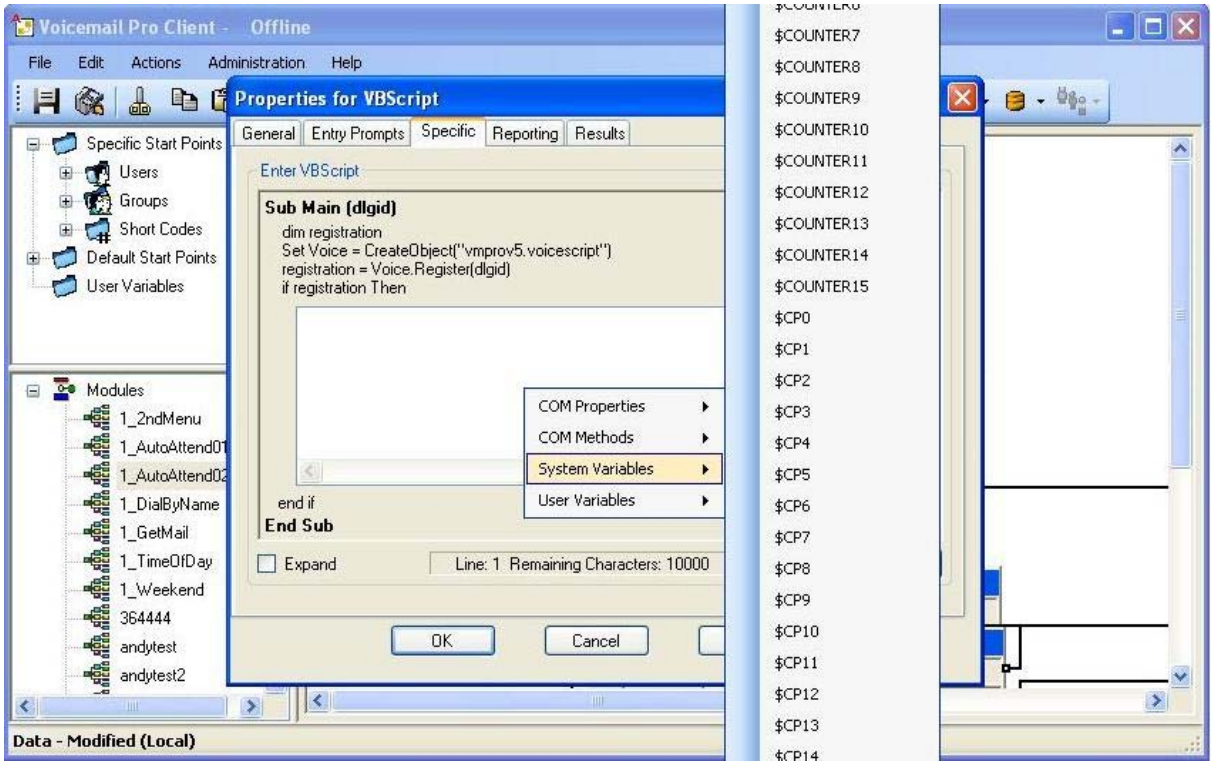
The Avaya TTS is a per-port license.

The total number of TTS ports on each VoiceMail Pro system is limited to eight. On a single or Centralized VoiceMail Pro, up to eight concurrent ports can be used for IVR and email reading, on a distributed VoiceMail Pro up to eight ports can be used for IVR.

Visual Basic (VB) Scripting

The VoiceMail Pro call flow programming interface allows an administrator to provide Visual Basic (VB) scripted logic that can be interpreted by the VoiceMail Pro server. This ability enables system administrators to program the voice system via VB Scripts thus providing additional choice and flexibility in providing IVR applications. The VB script action contains a VB-Scripting parser (Syntax checker) to ensure the legitimacy of the administrator derived VB Script before its incorporation. Each VB script action used within a call flow can contain a maximum of 10,000 characters; however a call flow may contain multiple VB script actions within it.

On a right mouse click in the VB Script window the methods and variables are available will be listed to assist when constructing call flows.



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VB Scripting is enabled by the IP Office Advanced Edition license.

Extended Personal Greetings

In INTUITY emulation mode, the VoiceMail Pro system has the ability to hold a number of greetings within each user's mailbox that can be played to a caller. In addition to the standard mailbox greetings, the extended personal greetings provide the ability to present the caller with a greeting that reflects where the call has come from (internal or external) or why the called party is unable to take the call. A mailbox user can configure the responses played back to the caller, based upon the reason the caller was routed to the voicemail. The supported call states are:

- Busy/Engaged
  The user is currently on a call and unable to accept a second call.
- No Reply
  The user is away from the desk and unable to take a call.
- Internal
  A greeting to be played to internal calls
- External
  The greeting to be played to external callers
- Out Of Hours
  The greeting played when a hunt group is operating 'out of hours'. Out of hours is defined with IP Office Manager and is only applicable to hunt group mailboxes.

A greeting can be recorded for each of the above conditions through the Telephone User Interface (TUI). If a recording is made for each condition, the order of play back to a caller will be:

1. Out of hours (Hunt group mailboxes only).
2. Internal/External greeting.
3. Busy/Engaged.
4. No reply.

A mailbox owner will need to record greetings against these conditions to deliver the greeting that they wish to present to a caller. one-X Portal users can record and manage their voicemail greetings through the one-X Portal user interface.

Hunt Group Broadcast Messages

With VoiceMail Pro, two modes of operation exist for the handling of hunt group messages. The method used is configured for the group through the IP Office Manager.

- Hunt group mode
  Messages are stored in the hunt group mailbox and Message Waiting only informs those individuals configured for message waiting indication from that group. This is ideal for scenarios where only a few people such as a call center supervisor need to be initially aware of group messages. Any message waiting light lit by this is extinguished when the new hunt group message is accessed by a user. This is the default mode of operation.
- Broadcast mode
  Messages are not stored in the hunt group mailbox. Instead they are broadcast (copied and forwarded) to the individual mailboxes of the entire hunt group membership. This lights the individual messages waiting light of each user of the hunt group until they access their mailbox.

Personal Distribution Lists

Personal Distribution Lists are only available with VoiceMail Pro when operating in INTUITY emulation mode. The feature provides the ability for a user to distribute a voicemail message to a list of recipients simultaneously. Lists can be configured by a voicemail box subscriber either through their voicemail box telephone user interface (TUI) or through the desktop PC application Phone Manager Pro.

The features available to a voicemail box subscriber include:

- Create up to 20 lists with 360 members per list
- Mark a list as private or public, Private lists can not be accessed by any other voicemail subscriber. Public lists can be used by other subscribers but can not be edited.
- Public lists can be copied from one subscriber to another by adding the contents into a new list.

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- Subscribers can 'Create' new lists, 'Scan' contents of an existing list or 'Modify' existing lists.
- List members can be added by using the station number or mailbox name (names are not supported for VoiceMail Pro Networked Messaging mailboxes).
- Lists can include voicemail boxes that exist on other Avaya voicemail systems that are available through VoiceMail Pro Networked Messaging.
- Lists can be added together, duplicate members are automatically removed. This includes public lists owned by other voicemail subscribers.
- Mailing lists are accessible to the user at any ‘send message’ and ‘forward message’ option within the user's voicemail box.
- When displayed within Phone Manager Pro, distribution lists can have a list description added to it, this is only visible within Phone Manager Pro.

Cascaded Out-Calling

VoiceMail Pro can send a notification, with an escalation capability, that a new voice message has been received in a user's mailbox to specified phone number(s). This is particularly useful in environments such as healthcare and support where important voice messages are left and need to be answered promptly - even outside of office hours.

For example should a patient leave an important message to the main number of the doctor’s office, the voicemail system can ring the doctor at the office then on no response escalates to the doctor’s mobile/cell phone, his/her home phone or the doctor on duty after a programmable timeout. This avoids having to rely on an external answering service and allows mobile/cell and home phone numbers to remain private.

The voicemail notification can be sent for:

- Any new voice messages
- Any new priority voice messages

Mailbox owners can configure their own options from their handset (Telephone User Interface or TUI) and through their IP Office Phone Manager.

- Create own Time Profile – defining when notification should take place (e.g. office hours only)
- Out-calling destinations – defining where notification should take place and in which priority order

Five destinations can be defined by the mailbox owner through the TUI (Telephone User Interface). The destinations selected in the escalation list are called in sequence. The possible destinations are:

- Desk
- Mobile/Cell
- Home
- Delegate
- Other

Each time an outcall event occurs, each number in the escalation list will be called until either the call is answered, or the end of the list is reached. This process will be repeated on each retry attempt, for the number of retries set.

Out-calling preferences are set for global operation via the VoiceMail Pro Client. Out-calling is only available in INTUITY Mode. The administrator sets the number of retries and time interval between retries on a system-wide level.

Campaign Manager

As part of the IP Office Preferred Edition and VoiceMail Pro, Campaign Manager enables the gathering of repetitive information from inbound calls (such as brochure requests) to be fully automated, leaving agents free to deal with other more complex calls which require human interaction. A definable sequence of recordings are played to the caller with time in between each recording to allow the capture of the caller's spoken answers and/or the caller's key presses via DTMF. At the end of the transaction the caller can be thanked and the completed transaction retrieved by an agent via a web interface or a short code.

Campaign Manager allows calls in queue to break out of the queue, or be directed in an overflow situation to complete their transactions thereby increasing customer satisfaction by affecting an answer to their call. This ensures that a minimum of customers give up when forced to wait in a queue or even worse, hear a recorded message stating that they are calling outside of office hours.

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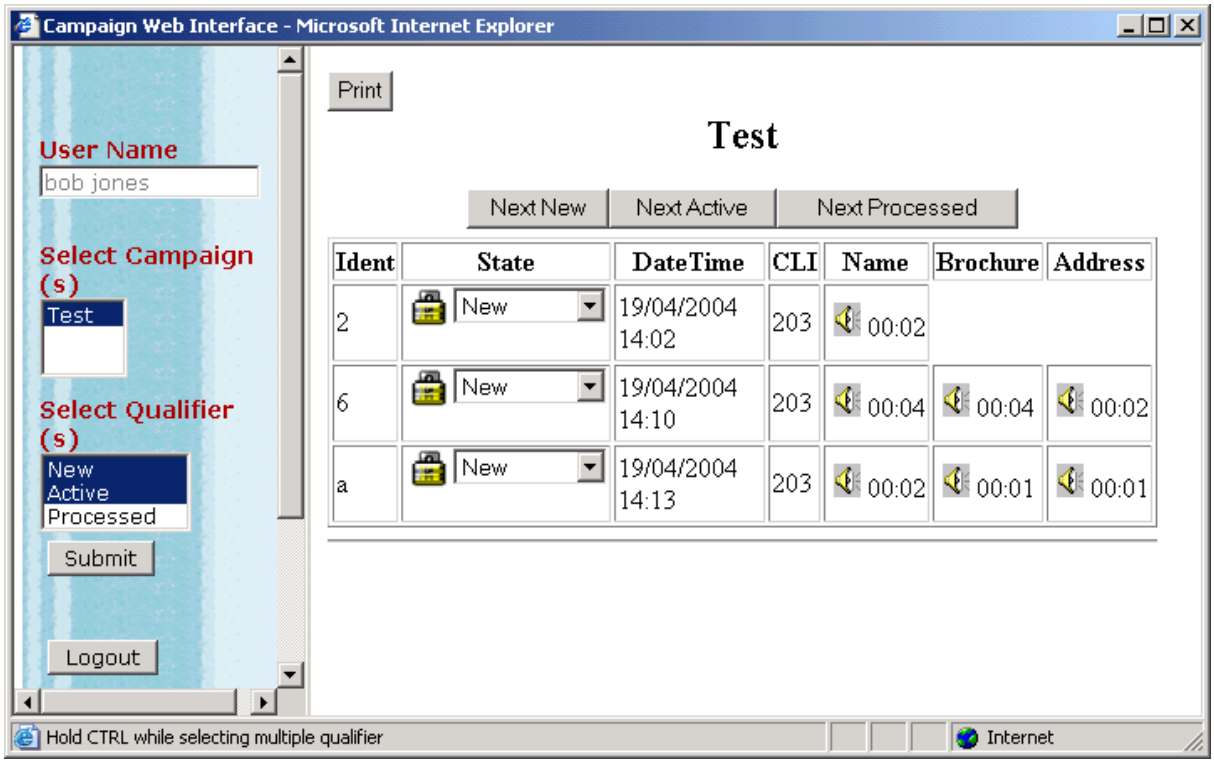
In a Contact Center environment, when agents are busy, an overflow to Campaign Manager relieves congestion and pressure on agent groups. An agent can collect the completed transaction via a web browser or via a short code representing the park slot number of a particular campaign. This number can be pre-programmed under a DSS key and used by agents to access the campaign. If the DSS key incorporates a BLF lamp, that lamp is lit when new campaign messages have been left. Agents then transcribe the caller's answers into a database or other records.



Call Recording

IP Office Preferred Edition and VoiceMail Pro also offer call recording services that allow the automatic/manual recording of calls for a variety of applications, such as for training purposes or to monitor abusive callers. As standard, recordings can be directed to the called extension's voicemail box or to any other mailbox for later retrieval. Alternatively, recordings can be stored in a central database for retrieval through a Web based browser by using ContactStore for IP Office.

An option allows to detect a tamper/ verified recording to avoid any abuse of recordings. This is available for recordings in a voicemail box and in the central database of contact store as well.

The system administrator can select whether all calls are required to be automatically recorded or just a selection of calls. Alternatively, calls can be manually selected for recording. If for any reasons resources are not available then a recording may not be taken (for example all voicemail ports are busy).

VoiceMail Pro provides a number of methods for triggering the recording of a call.

Most of the settings and controls for automatic voice recording are accessed through the IP Office Manager application. The proportion of incoming and/or outgoing calls that should be recorded and the time-period during which Voice Recording should operate can be selected.

- User Recording
  The calls to and/or from a particular user can be automatically recorded. By default the recordings are placed in the user's mailbox. The configuration could be set whether on "external" calls only (default) or on "external and internal" calls.
- Hunt Group Recording
  The calls to a particular hunt group can be automatically recorded. By default the recordings are placed in the hunt group's mailbox, but there is the ability to select a target mailbox made for or on behalf of a subscriber. The configuration could be set whether on "external" calls only (default) or on "external and internal" calls.
- Account Code Recording
  An account code can be applied to a call by the user before it is made. This can be used to trigger recording of outgoing calls.
- Caller ID Recording
  Account codes can be assigned to a call by Caller ID matching. This allows recording to be based on a Caller ID match.

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- Time Profiles
  For each user, hunt group and/or account code, an IP Office time profile can be used to determine when auto-recording is used.
- Incoming Call Routes
  Incoming Call Routes can trigger automatic call recording.

Note: It is possible for several recordings to be made of the same call. For example, if both automatic hunt group recording and automatic user recording are applicable to the same call, separate recordings are produced for both the hunt group and the user. Recording only continues while the party triggering the recording is part of the call, for example:

- Recording triggered by a user stops when that call is transferred to another user.
- Recording triggered by a hunt group continues if the call is transferred to another member of the same group.
- Recordings triggered by an incoming call route last until the call is cleared from the system.

Call recording uses the conference facility and so is subject to the conference restrictions of the IP Office system. For some situations, it may be a requirement that call parties are advised that their call is about to be recorded. This is done by switching on the Play Advice on Call Recording option via the VoiceMail Pro client. The maximum length of any call recording is 60 minutes.

IP Office ContactStore

The standard call recording facilities provided with IP Office and VoiceMail Pro can be extended further by using IP Office ContactStore. IP Office ContactStore stores and catalogs recordings so that they are easily accessible for later retrieval. Any recordings that you instruct VoiceMail Pro to "send to the Voice Recording Library" are placed in a database.

IP Office ContactStore is provided with the VoiceMail Pro software DVD set and has an inbuilt 45 day trial license. A fully featured IP Office ContactStore system can be installed and used for 45 days from the creation of the first recording. After this time the system will stop taking recordings until a license is purchased and installed onto the IP Office.

IP Office ContactStore has a number of components, these are:

- An MSDE database into which details of all recorded calls are inserted.
- A browser-based call search and replay application.
- A browser-based system configuration and status monitoring application.
- Disk space management - Oldest recordings are automatically deleted as needed.
- Optional archive management - Recordings are automatically written to a DVD +RW drive.

To allow you to search for calls easily, the details of the recordings are stored within a MSDE database. It contains one record for each call recorded and additional records for each party on the call and the owner of the call. The information that is held for any recording is:

- A unique reference for the recording
- The start date and time
- The duration of the recording
- The name and number of the parties on the call- where this was available to IP Office (through ANI, Caller ID or DNIS) at the time of the call.
- The direction of the call (incoming, outgoing, or internal)
- The owner of the call recording
- The target or dialed number, which may be different from the number that actually took the call.

Recordings within IP Office ContactStore are stored as .WAV files. IP Office ContactStore uses the G.726 16kbps ADPCM compression standard, which provides the best compromise between storage capacity and CPU loading. IP Office ContactStore is designed to perform compression as a background task, which does not impact the systems ability to record, search or play other calls. It takes approximately 1 minute to compress a two hour recording. The compressed recordings are stored as 16kbps G.726 format, storage requirements are therefore 8MBs per hour of recording.

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The IP Office ContactStore suite can be installed onto the same server as VoiceMail Pro but must be loaded onto a separate partition. Alternatively, IP Office ContactStore can be installed on a separate drive within the same server or on a separate server. The minimum PC specification when VoiceMail Pro and IP Office ContactStore are installed on the same server is detailed in the Voicemail System requirements later in this chapter.

IP Office ContactStore stores recorded calls with certain security in place. Access to recordings is strictly controlled according to the security constraints configured within the System Administration pages. Each recording has an owner; the call owner is the number of the extension that recorded the call. You can specify to which extensions each user has replay rights; the user can search for and replay all calls "owned" by those stations. Typically an individual may be given rights to replay calls owned by their extension number while managers may have rights to the extension numbers of all of their staff.

The system will automatically generate alarms showing system warnings. Alarms are logged to IP Office ContactStore's database and held for a month before being purged. The administrator can define specific Email addresses for alarms to be automatically forwarded to. The email recipient could be a local system administrator, a manned help-desk and/or suppliers' support desks if you have a support agreement that includes this facility. The system sends an email message each time an alarm occurs or is cleared. It also sends an email once per day as a "heartbeat" to let you know it is still operating. Failure to receive the daily heartbeat message should be investigated; it could indicate that the server has failed.

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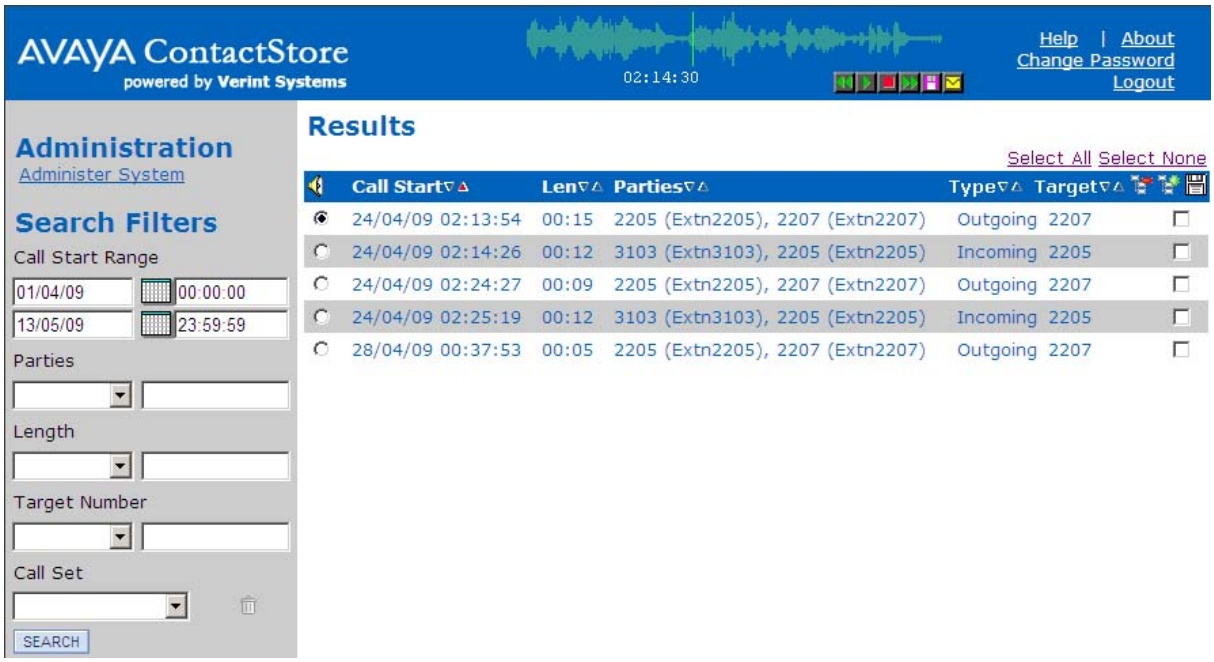
IP Office ContactStore allows replay of recordings by means of a browser-based application that is accessible with Internet Explorer (IE) V6.0 and higher. The Search and Replay facilities include the following features:

- Personal security restrictions. The restrictions are applied as you log into the web server.
- Criteria-based search filter fields to perform specific searches.
- Replay controls. Use the replay controls to start, stop, pause, skip forward, skip backward, or to export the recording to a readily playable .wav file.
- Audio waveform display. The waveform presents a graphic representation of the audio content of the call. Use the waveform to avoid replaying static or silences, and to move easily to specific portions of a call.

The Search and Replay screen, shown below, provides filter fields that you can use to search for calls:



For a single IP Office, ContactStore is enabled by the Advanced Edition license. Note: For ContactStore to work for remote nodes across a Multi Site Network there needs to be a ContactStore license present on each node. This is in addition to the Voice Networking license required for Multi Site Network.

Centralized Messaging with Avaya Communication Manager

Where IP Office is deployed in an Avaya Communication Manager (ACM) Environment, it may be desirable to use one centrally managed voicemail system (Modular Messaging) to provide voicemail services to IP Office users. IP Office can be configured to use a Modular Messaging system over a remote connection such that all messaging calls divert to this location and message waiting indications are provided from the remote location and are displayed correctly on IP Office extensions. Connectivity must be either an E1 or T1 circuit or an IP trunk running QSIG services. In addition to the IP Office license Key (Centralized VM with ACM RFA) that enables this service, further license keys may be required on the ACM system.

Voicemail Feature Comparison

Platform Support

	IP Office Preferred Edition VoiceMail Pro	IP Office Essential Edition Embedded Voicemail
IP500 V2	Yes	Yes
IP500	Yes	Yes

Capacities

Voicemail	IP Office Preferred Edition VoiceMail Pro	IP Office Essential Edition Embedded Voicemail

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Number of Mailboxes supported	No Limit - Limited only by IP Office configuration.	Limited only by IP Office configuration.
Maximum Number of Concurrent Calls (ports)	Up to 40 dependent on license	6 simultaneous calls on IP500 V2. 4 simultaneous calls on IP500.
Recording Time	PC dependent (Requires 1MB per minute)	IP500 V2: 2 ports: Up to 15 hours 4 ports: Up to 20 hours 6 ports: Up to 25 hours IP500: Up to 15 hours

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Features

	IP Office Preferred Edition VoiceMail Pro	IP Office Essential Edition Embedded Voicemail
Runs as a service	Yes	No
Multi-lingual support	Yes	Yes
Voicemail for Individual users	Yes	Yes
Voicemail for Virtual users	Yes	No
Voicemail for Hunt Groups	Yes	Yes
Group Broadcast	Yes	No
Unified Messaging Service (UMS)	Option	No
Integration with Microsoft Exchange Server 2007/2010	Option	No
Capable to interact with Blackberry solution	Option[1]	No
Resilience and Backup	Option	No
Small Community Network Operation	Yes	No
Centralized Voicemail Services	Yes	No
Distributed Voicemail Servers in an SCN	Yes	No
Voicemail Ringback	Internal and external	Yes
Voicemail Help TUI	Yes	No
Message Waiting Indication	Yes	Yes
Visual Voice (interactive menu on phone display)	Yes	Yes
Integration with Phone Manager Pro	Yes	No
Personalized Greeting	Yes	Yes
Extended personal Greetings	Yes[2]	No
Continuous Loop Greeting	Yes	No
Forward to Email	Yes	Yes
Copy to Email	Yes	Yes
Listen To Email (Text To Speech)	Yes[2]	No
Send Email notification	Yes	Yes
Unified Messaging Service (UMS)	Option	No
Save Message	Yes	Yes
Delete Message	Yes	Yes
Forward Message to another Mailbox	Yes	Yes
Forward to Multiple Mailboxes	Yes	Yes
Forward with a Header Message	Yes	Yes
Repeat Message	Yes	Yes
Rewind Message	Yes	Yes
Fast Forward Message	Yes	Yes
Pause Message	Yes	No
Skip Message	Yes	Yes
Oldest message first/newest message first Message Playback Option	Yes	No
Set Message Priority	Yes[2]	No
Set automatic message deletion timeframe	Yes	No
Alphanumeric Data Collection	Yes[2]	No
Callers Caller ID, time and date announced	Yes	Yes
Call Back Sender (if Caller ID available)	Yes	Yes
Remote Access to Mailbox	Yes	Yes
User Definable PIN Code	Yes	Yes
Known Caller ID PIN Code By-Pass	Yes	Yes
Breakout to Reception	Internal and external.	Internal and external.

1. Requires UMS (enabled through the Power User, Office Worker and the Teleworker licenses) and MS Exchange Server 2007/2010 with a mobility solution (for example a Blackberry) - not provided by Avaya.
2. Intuity mode only.

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In-Queue Announcements

	IP Office Preferred Edition VoiceMail Pro	IP Office Essential Edition Embedded Voicemail
Queue Entry Announcement	Yes	Yes
Queue Update Announcement	Yes	Yes
Queue Position Announcement	Yes	No
Time in Queue Announcement	Yes	No
Time in System Announcement	Yes	No
Estimated Time to Answer (ETA)	Yes	No
Exit Queue to alternative answer point	Yes	No

Auto-Attendant/Audiotex

	IP Office Preferred Edition VoiceMail Pro	IP Office Essential Edition Embedded Voicemail
Multi-Level Tree Structure	Yes	Yes
Message Announcements	Yes	No
Whisper Announce	Yes	No
Alarm Calls	Yes	No
Assisted Transfers	Yes	No
Dial by Name	Yes	Yes
Direct Dial by Number	Yes	Yes

Other Features

	IP Office Preferred Edition VoiceMail Pro	IP Office Essential Edition Embedded Voicemail
Call Recording	Yes	No
Tamper proofed / verified Call Recording	Yes	No
Test Conditions	Yes	No
Personal Numbering	Yes	No
Speaking Clock	Yes	No
Campaign Manager	Yes	No
VoiceMail Pro Manager	Yes	No
Customized Voicemail	Yes	No
Intuity TUI emulation mode.	Yes	No
Forward Emails to External Systems (VPIM)	Yes	No
Third Party Database Access (IVR)	Yes	No
Text To Speech within call flows	Yes	No
Support for Visual Basic Scripts	Yes	No

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Chapter 14. Mobility

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14. Mobility

Sales or service reps who are always on the road know the challenges: missed calls, phone tag, languishing voice messages, delays in decision making and frustrated customers. Plus the real possibility of missed transactions and lost revenue. Equipped with Avaya IP Office Mobility Solutions, mobile associates never need to miss an important customer call or a question from someone in the office. And they'll be able to give customers just a single phone number — their office number — where they can be reached regardless of their location. Personal mobile phone numbers are never given out, but all calls ring simultaneously on their desk telephones and their mobile telephones. So if they are at their desks or on their mobile telephones, they'll get the call. With IP Office Mobility Solutions they maintain the agility that gives them an edge over larger competitors. Decision-making will be speeded-up, quick responds to customers and colleagues are enabled, and the solutions will help to ensure that deals close with this real-time connection.

On-Site Mobility

IP Office offers different solution for users working mainly at the business campus but with the need to be accessible all over the site.

Avaya On-Site Wireless Phone Solutions

Avaya IP Office Wireless Solutions include DECT and Wi-Fi based wireless IP telephones. These are solutions employees can use every day to work more effectively and be more responsive to customers - all while increasing revenues and keeping communication costs firmly under control. Also, Avaya IP Office Mobility Solutions integrate seamlessly with IP Office, enhancing each customer's investment. IP Office's in-building Mobility Solutions improve communication with staff who, because of the function they perform, are mobile within the workplace. Using wireless technology, such individuals may be instantly contactable, with many obvious benefits:

- The wireless telephone is carried in the pocket, so users are not tied to the desk in order to remain in contact.
- Users may be contacted instantly to ensure fast, accurate decision making and immediate response to problems through planned radio coverage with no blind spots.
- Once installed the solutions do not incur additional cost like mobile telephones would do, allowing a tight budget control.

Wireless Solutions

IP Office supports the following wireless solutions:

- DECT R4, the DECT based solution for IP Office. Running in a separate, reserved frequency range, this solution provides best in class voice quality and is totally interference free from other radio equipment. Details on this solution portfolio can be found in the chapter "Telephones" of this document.
- Avaya VoIP Wi-Fi Solution, leveraging the Wi-Fi data network, allow a single infrastructure for both voice and data traffic. Leveraging an existing, voice ready Wi-Fi network allows a cost efficient deployment of a converged wireless data and voice infrastructure using the 802.11a, b and g standard. The solution is offered worldwide in selected countries including North America, EU countries and selected others. For details on this solution please refer to the chapter "Telephones" of this document.

Remote Solutions

IP Office supports the following VPN remote phone solutions:

- VPN phone client on 4610SW, 4621SW, 5610SW, 5621SW and 9600 Series IP telephones offered worldwide.

For further details about IP Office On-Site Mobility solutions please refer to the chapter Telephones 98 of this document.

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Twinning

Twinning allows a primary extension and a secondary number (internal extension or external number) to operate together as a single telephone, when a call is presented to the primary phone the secondary will ring. If the primary telephone does not ring, for example in Do Not Disturb, the secondary phone will not ring.

The following types of calls are eligible for twinning:

	Internal Twinning	Mobile (external) Twinning
Any internal call on a Call Appearance button	✓	✓
Internal or external calls transferred to the extension	✓	✓
Direct Dial calls to that extension	✓	✓
Hunt group calls	✓	✓
Calls forwarded from another extension	✓	✓
Line Appearance calls (configurable)	✓	✗
Bridged Appearance calls (configurable)	✓	✗
Coverage calls (configurable)	✓	✗
Automatic Intercom calls	✓	✗
Returning transferred, held or park calls	✓	✓*
Callback calls from the system (Transfer and Park Return)	✓	✗
Paging calls	✓	✗
Follow Me calls	✓	✗

*Requires IP500 BRI, PRI or SIP trunks and Mobile Call Control to be enabled for that user.

Internal Twinning

This is typically used in scenarios like workshops or warehouses where team supervisors may have a desk with a fixed phone but also have a wireless extension (e.g. DECT). Other users of the system need not know that the supervisor has two different telephones. The supervisor's Coverage Timer and No Answer Time are started for the call and if the call is not answered within that time, the call will be delivered to available coverage buttons (if applicable) and then Voicemail (if applicable).

The following features are supported with internal twinning

- Follow Me To
- Follow Me Here
- Forwarding
- Do Not Disturb (inc exceptions)
- Context less hunt group actions: Membership / Service Status / Fallback Group configuration
- Voicemail On / Off / Access
- Call Log (Central Call Log for T3 and 1400/1600/9600 telephones only)
- Redial (Central Call Log for T3 and 1400/1600/9600 telephones only)
- Personal Directory Entries (for T3 and 1400/1600/9600 telephones only)

External Mobility

Mobility functions for employees working not from the office but mobile are enabled either by the IP Office Power User or the IP Office Mobile Worker license. The former Mobile Twinning / Mobility License will continue to work with all the features described below.

The mobility functions enabled by these user licenses are Mobile Twinning, and Mobile Call Control, as well as one-X Mobile client support. Each feature has to be enabled in IP Office Manager for eligible users.

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Mobile Twinning (External Twinning)

A typical use for Mobile Twinning would be for a salesperson that has a desk extension but is often out of the office. They only give out their office phone number to clients, but the IP Office Mobile Twinning will ring their mobile/cell phone as well as the desk extension. External mobile devices can be set as twinning targets even if the primary telephone is logged out (e.g. in the case of a "virtual" extension).

Call Back

To avoid any call charges when using a mobile / cell phone the call back feature has been implemented:

When calling a particular number at the IP Office, the IP Office will reject the call and will call back the mobile number if this number has been identified as an eligible one. After accepting this call the mobile user gets a dial tone and can establish another call. Depending on the service provider's tariff, there are no call charges (except possible roaming charges) on the mobile invoice. However, in some countries like US there are charges to even receive the calls.

This feature is very useful for employees using their personal mobile phone for business only from time to time: their personal mobile phone number stays confidential and they do not to claim mobile charges back from their business.

Avaya one-X Mobile Client Support

IP Office supports the Avaya one-X Mobile 'Single Mode' Clients running on selected Symbian Single-Mode or Windows Mobile 5 or 6.x handsets. This provides a graphical interface for call control and allows a one-number service for both incoming and outgoing calls.

For a full list of supported handsets see http://support.avaya.com.

IP Office systems supporting one-X Mobile require a dedicated DID/DDI number for each feature required (e.g. hold, transfer).

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Mobile Call Control

This allows a user who has answered a twinned call on their mobile/cell phone to dial a DTMF sequence ** (star-star) to put the call on hold and receive IP Office system dial tone. The user will then be able to perform supervised and un-supervised transfers, shuttle (switch between held calls), and conference (Conference add and transfer to a meet-me conference).

Mobile Call Control is a mobility feature similar to but independent of one-X Mobile. It allows any mobile phone user to access IP Office features without using the one-X Mobile client application.

Mobile users can gain access to the “Star Star Service” dial tone when they receive a twinned call. After receiving a twinned call, the mobile user can dial ** to place that call on hold and get dial tone from the IP Office. Once the mobile user receives IP Office dial tone, any subsequent dialing done is interpreted as if the user is logged into an analog extension on the IP Office.

To gain access to the “Star Star Service” dial tone, the mobile user can dial the FNE 31 DID/DDI that has been programmed in the IP Office. Similarly, once dial tone is heard, the user can dial numbers and short codes as if the user is logged into an analog extension on the IP Office.

Mobile Call Control and one-X Mobile require twinned calls to be routed through specific trunk types on IP500 / IP500v2 only:

- IP500 PRI-U (single or dual) trunk cards
- IP500 Dual or Quad BRI trunk cards
- SIP trunks (RFC 2833)

IP Office Softphone

Usable both inside the office and at remote locations, IP Office Softphone is a PC-based telephony client allowing a user to make and receive phone calls right from his PC, equipped with e.g. a headset of speakerphone. In addition, the softphone supports end to end Video between two softphone users on a IP Office network. Full voice access to the IP Office system is available wherever reliable access to the company network exists.

This can be in meeting rooms, using a Voice ready Wi-Fi network, at a remote office site using the local Ethernet interface or at home or in a hotel-room, when connecting to the company network through a VPN (Virtual Private Network).

The softphone offers standard calling functionality, like several call appearances, hold, transfer and access to a large number of IP Office features via dedicated feature keys. Access to IP Office directory is of course also available.

IP Office Softphone can be used as a standalone client (not requiring any additional server) or in combination with one-X Portal for IP Office.

If used in combination with the one-X Portal application, the softphone delivers a centralized call log, synchronized with many IP Office telephones as well as a number of collaboration features like presence status indication and instant messaging. one-X Portal can be used with the standard deskphone while working in the office and switched to softphone when working remote.

Details about IP Office Softphone can be found in the chapter User Call Control Applications 274.

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Mobile Messaging

Mobile Messaging means managing your voice messages together with email messages using the mobile / cell phone. Similar to a unified messaging solution on the desktop PC it enables the user to manage all of their daily messages in one place, in this case the mobile messaging application of the mobile / cell phone (like for example Outlook Mobile on Windows Mobile devices).

At the low end of the range of mobile messaging options is the simple voicemail-to-email forwarding using SMTP, which is standard with both the Essential and Preferred Edition.

The synchronization of the voice message status between all a user's devices (Outlook, voicemail telephone interface, one-X Portal, mobile / cell phone) so that listening to a new message at one device changes the message status from "new" to "old" in all of these interfaces requires the Preferred Edition. It contains an IMAP interface to integrate the VoiceMail Pro solution with any IMAP capable email system.

With the IP Office Preferred Edition in conjunction with an Exchange 2007/2010 Server it is possible to synchronize received voice mails with the email system. If the email system also empowers mobile users to receive, manage, and send messages by their mobile phone ('Mobile Email') all voice mails can be managed as well from the mobile / cell phone.

Together with an appropriate mobility solution like Blackberry the Exchange 2007/2010 integration enables visual voicemail on mobile devices. The voice mails provided by the Preferred Edition are specially marked as "voicemail" so Exchange 2007 handles them different from standard emails so that a mobility solution will be able to show voice mails on the display of the mobile device. Users can now browse through the voicemails, identify high priority ones by caller information and listen to them first.

3rd Party Mobility Solutions for Use with IP Office

Although Avaya offers different mobility solutions as mentioned above there are areas for further improvement. Together with Avaya DevConnect partners Avaya is anxious to further extend the usability and functionality of the IP Office system.

The following mobility solutions in particular add mobile devices currently not supported by Avaya for the IP Office so that users with one of these devices can participate from the rich IP Office mobility options.

Apple iPhone – iEC5xx

As the iPhone from Apple is increasingly used for business use, Avaya has enabled an available DevConnect solution to extend IP Office mobility also to this popular device. The application for the iPhone is called 'iEC5xx' and acts like the Avaya one-X Mobile clients. The same configuration file that is been used to configure the Avaya clients enables the mobility functions at the iPhone.

The application is available at the Apple App Store.

For more information see http://elinfierno.de/apps/iec5xx/

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Blackberry – SoLo DTMF Client

One of the most used mobile business devices is the RIM Blackberry. To use a Blackberry with similar functions like the Avaya one-X Mobile application, the solution called “SoLo DTMF client” from SoSoftware enables selected Blackberry devices to use the IP Office mobility functions.

For more information and placing orders see
http://www.sosoftware.com/solo-mobile-dtmf

These 3rd party solutions have the same requirements as the Avaya mobility options. Each user who wants to use one of the solutions requires at least one Power User or Mobile Worker license. In addition, at least a Preferred Edition might be required.

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Chapter 15.
Audio Conferencing

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15. Audio Conferencing

A problem familiar to any organization is that of communicating effectively. As more and more people work from home or from dispersed locations, how do you ensure that employees are planning and working together effectively, and regularly keeping in touch when separated by time and distance? In addition, many companies choose to sub-contract some services such as payroll, logistics or manufacturing to third-party suppliers. How do you ensure that you can act as one virtual enterprise? Audio conferencing provides a simple and effective solution.

Audio conferencing makes it easy to include key people in decision making wherever they are with minimum interruption from their work. It responds to business needs that every company faces:

- More meetings but less time available.
- Increasing pressure to be at two locations at once.
- Travel restrictions.
- Environmental pressures ("green" initiatives).

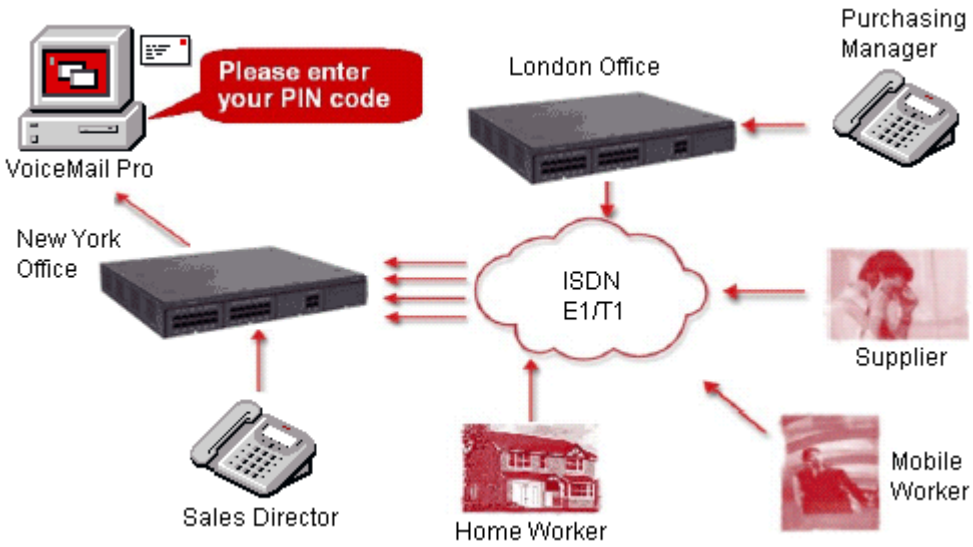
As a result of using conferencing, the benefits gained are:

- Reduction in travel, leading to lower costs, less wasted time and lower carbon emissions.
- Increased worker productivity and personal security.
- More effective working practices, leading to shorter project times, and supporting dispersed organizations and complex supply chains.

Furthermore, the Return On Investment (ROI) is very short as Meet Me conferencing is a built-in feature of IP Office Preferred Edition. The typical ROI of just 4 to 6 months compared to Service Provider conferencing services based upon 2 hourly conferences with 5 participants per week.

IP Office Preferred Edition Meet-Me Conferencing Solution

The conferencing solution built-in to IP Office enables multiple callers to talk in an audio conference. Callers can be on-site personnel as well as external parties whether field-based engineers, sales staff on the road, customers or suppliers. Conference calls can be planned in advance or established ad-hoc as and when required.



IP Office Preferred Edition with VoiceMail Pro complements the built-in conference bridge facility on IP Office systems by allowing participants to enter conferences through dedicated numbers, adding guidance prompts as well as requesting PIN codes as participants enter the conference for security. For example, if conference calls are regularly scheduled, VoiceMail Pro can have pre-programmed Call Flows for weekly conference calls e.g. every Tuesday between 2pm and 5pm using PIN code 1234 for a sales call, etc. If multiple conference calls are scheduled, users can select which one they want to attend via a simple menu. Should users encounter any issues, calls can be automatically routed to the operator for assistance. For additional security, if Caller ID information is provided by the network VoiceMail Pro can make CallerID checks before allowing calls into a conference.

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IP Office Conferencing Capacity

IP Office provides a flexible conferencing solution with 128 conferencing channels on the IP500 and IP500 V2, allowing multiple conferences of any size from 3 to 64 parties.. This means that several conferences of different sizes can all run at the same time if the total calls do not exceed the systems conference resources. So the system will support 42 x 3-party conferences, 2 x 64-party conferences or any combination in between. IP Office does not impose limits on the mix of internal and external calls in conference, but if all internal participants disconnect from the conference bridge, the external participants can be disconnected automatically by the system for added security (configurable system setting).

Notes:

1. Analog Trunk Restriction
   In conferences that include external analog line calls, a maximum of two analog line calls are allowed per conference.
2. External Participants
   Each external caller requires a digital trunk/VoIP channel (for example 1 T1 allows 23/24 external parties, 1 E1 allows 30 parties and a fully licensed VCM-64 allows 64 parties).
3. Use of Conference Resources by Other Features
   System features such as call intrusion, call recording and silent monitoring all use conference resources, as does automatic recording if enabled. When any of these features are active the number of slots available for conference parties is reduced. For example, a conference call between 3 parties and being recorded will use 4 conference slots.
4. The IP500 Supports 128 Conference Members
   The total capacity of the IP500 V2 is 128 party conference resources. However the restriction of a maximum of 64 parties in any individual conference still applies.
5. Meet-Me Conferencing on IP500 requires Preferred Edition
   IP Office IP500 V2 supports basic or ad-hoc conferencing, but if Meet-Me capabilities are required the Upgrade License to IP Office Preferred Edition should be purchased for direct dial into a conference bridge with PIN code security.
   Each IP Office unit which should host Meet-Me capabilities need their own Preferred Edition.

IP Office Standard Conferencing Features

The IP Office provides the following features and benefits relating to conferencing:

- No special conferencing equipment required
  You only need an IP Office system unit with as many digital trunks/VoIP channels as external participants (as well as Preferred Edition for Meet-Me conferences).
- Ease of use
  Simply dial the direct number allocated to the conference bridge, type in the PIN if required and you have joined the conference (PIN codes require Preferred Edition/VoiceMail Pro).
- Conference control from Avaya Display telephones*, IP Office Phone Manager and Avaya one-X™ Portal for IP Office
  For ad-hoc conferences with a few participants, users can easily set up immediate conferences by calling all parties and bringing them to the conference bridge. With Avaya one-X™ Portal for IP Office , the originator of the conference can keep control: the Caller ID number (and the associated name if recognized) of each participant is displayed. If required, they can selectively hang-up a specific participant.
- Customized greeting
  Record a personalized greeting per conference (requires Preferred Edition/VoiceMail Pro).
- Conference entry/exit tones
  Single beep on entry/double beep on exit
- Conference call recording
  Manual recording initiated by user on IP Office via Avaya one-X™ Portal for IP Office , Phone Manager, digital/IP display phone or a short code (requires Preferred Edition/VoiceMail Pro)
- Security
  To prevent unauthorized access to the conference bridge, PIN codes, Caller ID number screening as well as time and date profiles can be set-up using IP Office VoiceMail Pro.
- Privacy
  In cases where the security of calls is critical, in-house conferencing is the only way to ensure privacy.
- Remote Management
  Allows a single person to manage the conferencing bridge facility from any location. Furthermore, the full IP Office solution - phone system, voicemail, CTI server, router, firewall and DHCP server- can all be managed from a single management interface called IP Office Manager.

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*Feature supported on 1408/1608/2410/5410/4610/5610 and 1416/1616/2420/5420/5621/4621/4625 as well as 9600 Series IP telephones. Note that any internal party has the option to view & drop participants (not just the conference originator).

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Chapter 16.
The Contact Center

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16. The Contact Center

Avaya provides Customer Contact solutions that meet the needs of the small to mid-size business. From the smallest company that requires basic system performance reporting to the larger businesses that need advanced routing and multimedia integration with the Customer Contact Center. Avaya provides IP Office Customer Call Reporter as the perfect solution to meet the needs of small business customers.

Customer Call Reporter

Overview

Contact Centers are a requirement at nearly every level of business, from very small informal groups of 5 or 10 "customer service reps", to large formal campaigns that can involve thousands of agents.

However, customers in the small business market are generally very nervous about making such a high profile investment in a "contact center solution." They require a "business intelligence solution" that deploys easily, can be managed with very little training, and can provide statistics on each segment of the business. Small businesses also do not have the resources for extra IT staff to administer databases, servers, etc, so they need a solution that can be easily understood by their own employees. It should operate as a service to each of the clients.

IP Office Customer Call Reporter Concept

IP Office Customer Call Reporter is a server based contact center product designed explicitly for small businesses. Drawing upon the latest web and design technologies, IP Office Customer Call Reporter introduces significant new capabilities to effectively and efficiently manage a single-site call center environment. The product focus is on ease of use and deployment simplicity to ensure improved ongoing productivity and lower support costs.

In its native web server environment the IP Office Customer Call Reporter can be simply installed onto a single server without the need for any client software deployment. Administration and management of the call center is carried out via a thin client through a secure password protected web browser session.

IP Office Customer Call Reporter Key Features

- Real time Call Center activity monitor and historical reporting.
- Microsoft server based with thin client (web browser) client connections.
- Simplified installation and maintenance.
- Supervisor Dashboard which consists of a goal, statistics information ticker and a set of graph display panels.
- Seven reporting templates provide drag and drop and user defined filter functionality.
- Three customizable Supervisor views and summarized Agent views.
- Report Scheduler.
- Customer Map.
- Multi-lingual capable.
- Supports up to 30 Supervisors and 150 Agents.

IP Office Customer Call Reporter Business Benefits

- Lower TCO
  IP Office Customer Call Reporter provides small businesses with a web based contact center measurement tool, producing cradle to grave reporting in an easily understandable format with no client software to load.
- Standards Based
  IP Office Customer Call Reporter uses standards based applications, such as Microsoft SQL 2005 Express, and supports all major web browser software, to provide small business installers with greater flexibility in deployment.
- Ease of Use
  IP Office Customer Call Reporter's real-time charts can be customized by the user in the manner that suits their business best, with historical reporting templates that allow the business to filter on the type of date they want to see.

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Ordering IP Office Customer Call Reporter

Customer Call Reported supports the following maximum configurations for IP Office:

- 150 Agents
- 30 Supervisors

Activation of IP Office Customer Call Reporter is provided with the purchase of the IP Office Advanced Edition, the Advanced Edition is a specialized call center bundle which also enables the following:

- Contact Store for IP Office (CSIPO).
- Voicemail Pro database interaction (IVR).
- Voicemail Pro Visual Basic Scripting.
- 1 Customer Call Center Supervisor

Additional agents and supervisors can be purchased on top of the Advanced Edition

The IP Office has a setting in IP Office Manager for designating which hunt groups will be reported upon by IP Office Customer Call Reporter. Any hunt group that contains an agent who is not properly licensed will not be able to retrieve any reporting on that group (real-time or historical).

IP Office Customer Call Reporter Server Operating System Requirements

The Customer Call Reporter server is supported on the following platforms:

- Microsoft Windows Server 2008 (All Versions)
- Microsoft Windows Business Server 2003 R2.
- Microsoft Windows Server 2003 SP2.
- Microsoft Windows Server 2003 R2.

IP Office Customer Call Reporter Server Hardware Requirements

The minimum server specifications are as follows:

- IP Office Customer Call Reporter server
  Intel Pentium D 945 core/AMD Athlon 64 4000+; with 2GB RAM and 30 GB free hard disk space.

  Note: The IP Office Customer Call Reporter server can co-reside with a VoiceMail Pro of less than 17 ports.

IP Office Customer Call Reporter Client Components

The client application consists of a secure thin client web browser session to the IP Office Customer Call Reporter server that will provide the Supervisor or Agent with a complete view of the Call Center.

The following web browsers will be supported:

- Microsoft Internet Explorer 7 and above. Internet Explorer 8 recommended.
- Mozilla Firefox, version 3.0 and above.
- Windows Safari 3.1 and above.
- Apple Safari 3.1 and above.

IP Office Customer Call Reporter Database

The database has been designed to provide a simple table schema. The database is a Microsoft SQL Server 2008 instance and the IP Office Customer Call Reporter will be packaged with the Microsoft SQL Server 2008 Express edition. It is recommended that any call center that predicts high call usage should understand the imposed limitations of Microsoft SQL 2008 Express edition and consider using the full version of Microsoft SQL 2005 or 2008. Microsoft SQL 2000 is not supported.

The CCR Database is documented and open for third parties to create their own custom reports in Crystal Reports or any other SQL related application.

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Dashboard

This is the default page shown when a supervisor logs in. It shows information graphs for selected queues, agents and statistics.

Historical Reporting

The IP Office Customer Call Reporter reporting package has been simplified to enable the user to easily retrieve the reporting information they need from one of seven standard drag and drop report templates. The IP Office Customer Call Reporter offers "cradle to grave" reporting for businesses that want to track specifically how a call was routed through their business. The historical reporting is accessed via the Supervisor login.

IP Office Customer Call Reporter Report Templates

The following main report templates are available with IP Office Customer Call Reporter:

- Agent Summary Report
  This report type details the call activity for agents in a queue or for specific agents.
- Call Details Report
  The Call Details Report contains details of the total number of calls, internal calls (optional), external outbound calls and the total number of external inbound calls.
- Call Summary Report
  The Call Summary Report contains details of the total calls presented and answered; the average speed to answer in percentage and average speed to answer time, the number of calls overflowed and lost, the average abandon time as well as the grade of service.
- Trace Reports
  The Trace Report lists in chronological order all the events, date and time stamped that involved the user selected.
- Alarm Reports
  Alarm reports can be created for the Supervisor Views. All the alarms for the selected subject are listed over a user specified time period. Totals are included that show the number of warning and critical alarms received for each unique alarm type included in the report.
- Voicemail Reports
  The report shows the following information; Total calls answered by the Auto-Attendant (VMPro), total calls abandoned (lost) in Auto-Attendant, and a table of unique call flow trigger names with a tally of the number of times the trigger has been received (e.g. who pressed 1 for sales in the Auto-Attendant).
- Agent Time Card Report
  This report summarizes agent performance including time logged in, talk time, time in various states, calls made and answered and agent performance. The report does not include internal calls.

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Creating an IP Office Customer Call Reporter Report

From the seven basic report templates within IP Office Customer Call Reporter, a user can create over 100 different reports depending upon how they want to see the information that is most useful for their business. All standard report templates have common fields that can be completed to create a report. And when entering details in a field a wildcard '*' can be used to represent everything. The common fields are listed below:

- Report Name
  The text entered is used as the main title of the report. If the report is saved this is the saved name of the report.
- Subject
  This is the main subject of the report. The drop down list offers a list of report subjects that are pertinent to the report i.e. Hunt Group, Agent, DDI etc.
- Filter
  The filter is used to refine the data and again offers filters that are pertinent to the report i.e. All, Answered, Lost, etc.
- Exclude Internal
  When selected, internal calls are excluded from the report.
- Working Hours
  The start and finish time of the working day can be specified using the spin controls.
- Group Results by
  The drop down list offers a list of display options.
- Scheduled
  By default this field is shown as 'Manual'. If the option 'Scheduled' is selected then additional fields are available:
- Report Format
  The format that the report is to be produced in can be selected from Adobe, Word, Excel, Rich Text or Crystal.
- Email Address
  An email address can be entered if the report is to be sent by email.
- Printer
  The printer path that the report should be sent to.
- Report Period
  Selection of days, weeks or months from a list box. The number of days or weeks is counted from the date that the report is first scheduled to be run. Months are from the last full months data i.e. if the report is scheduled in February then January will be the last month in the report.
- Report Range
  The start and end date for the report.
- Language
  If the default language is not required then another language can be selected that your system supports.

Filters can be used in the report to provide exactly the type of information the customer is looking for. The filter field is used to refine the data and offers the following options in a drop down list; All, Answered, Refused, Refused Overflowed, Overflowed All, Overflowed Off PBX, Overflowed on PBX, Transferred and Lost Calls. The default setting is all.

Reports can also be scheduled for future delivery to any network printer or email address in any of the supported formats.

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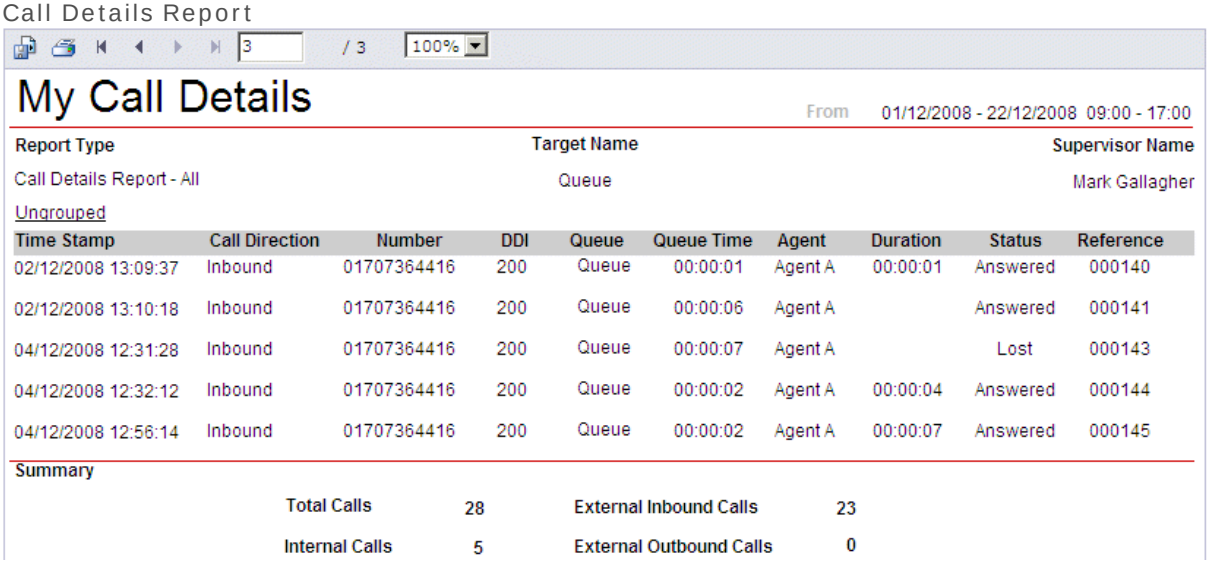
Call Details Report



My Call Details

From 01/12/2008 - 22/12/2008 09:00 - 17:00

Report Type	Target Name	Supervisor Name
Call Details Report - All	Queue	Mark Gallagher

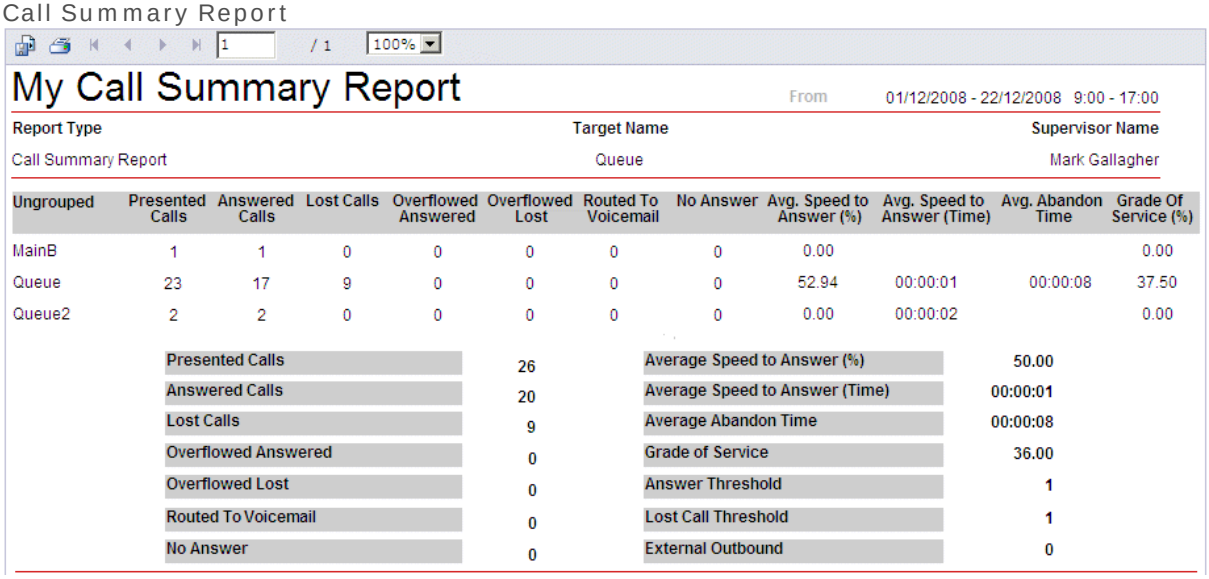
Ungrouped

Time Stamp	Call Direction	Number	DDI	Queue	Queue Time	Agent	Duration	Status	Reference
02/12/2008 13:09:37	Inbound	01707364416	200	Queue	00:00:01	Agent A	00:00:01	Answered	000140
02/12/2008 13:10:18	Inbound	01707364416	200	Queue	00:00:06	Agent A		Answered	000141
04/12/2008 12:31:28	Inbound	01707364416	200	Queue	00:00:07	Agent A		Lost	000143
04/12/2008 12:32:12	Inbound	01707364416	200	Queue	00:00:02	Agent A	00:00:04	Answered	000144
04/12/2008 12:56:14	Inbound	01707364416	200	Queue	00:00:02	Agent A	00:00:07	Answered	000145

Summary

Total Calls	28	External Inbound Calls	23
Internal Calls	5	External Outbound Calls	0

Call Summary Report



My Call Summary Report

From 01/12/2008 - 22/12/2008 9:00 - 17:00

Report Type	Target Name	Supervisor Name
Call Summary Report	Queue	Mark Gallagher

Ungrouped	Presented Calls	Answered Calls	Lost Calls	Overflowed Answered	Overflowed Lost	Routed To Voicemail	No Answer	Avg. Speed to Answer (%)	Avg. Speed to Answer (Time)	Avg. Abandon Time	Grade Of Service (%)
MainB	1	1	0	0	0	0	0	0.00			0.00
Queue	23	17	9	0	0	0	0	52.94	00:00:01	00:00:08	37.50
Queue2	2	2	0	0	0	0	0	0.00	00:00:02		0.00

Presented Calls	26	Average Speed to Answer (%)	50.00
Answered Calls	20	Average Speed to Answer (Time)	00:00:01
Lost Calls	9	Average Abandon Time	00:00:08
Overflowed Answered	0	Grade of Service	36.00
Overflowed Lost	0	Answer Threshold	1
Routed To Voicemail	0	Lost Call Threshold	1
No Answer	0	External Outbound	0

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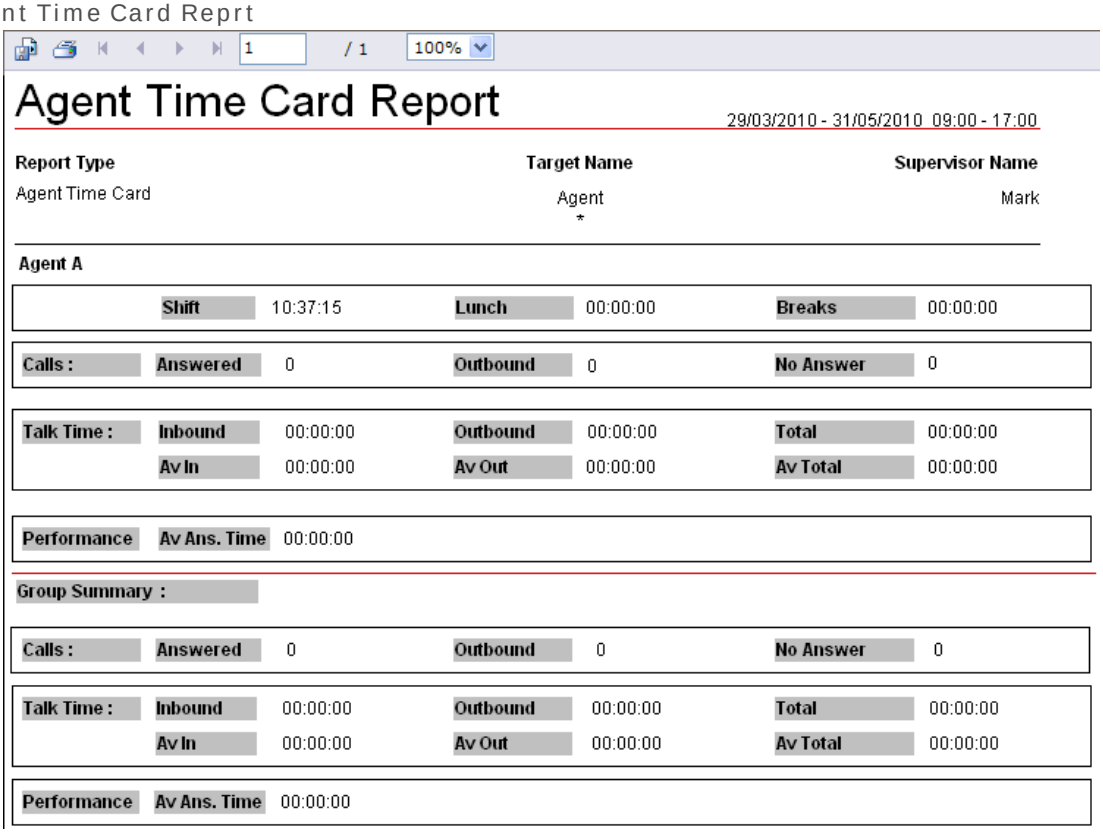
Agent Time Card Reprt



Agent Time Card Report

29/03/2010 - 31/05/2010 09:00 - 17:00

Report Type	Target Name	Supervisor Name
Agent Time Card	Agent *	Mark

Agent A

	Shift	10:37:15	Lunch	00:00:00	Breaks	00:00:00
Calls :	Answered	0	Outbound	0	No Answer	0
Talk Time :	Inbound	00:00:00	Outbound	00:00:00	Total	00:00:00
	Av In	00:00:00	Av Out	00:00:00	Av Total	00:00:00
Performance	Av Ans. Time	00:00:00				

Group Summary :

Calls :	Answered	0	Outbound	0	No Answer	0
Talk Time :	Inbound	00:00:00	Outbound	00:00:00	Total	00:00:00
	Av In	00:00:00	Av Out	00:00:00	Av Total	00:00:00
Performance	Av Ans. Time	00:00:00				

Supervisor Wallboard

For the release of IP Office R6, Customer Call Reporter now provides a browser based wallboard that can be used in businesses who want to display key statistics in a group area, such as on an LCD television screen.

The wallboard has the ability to display all the statistics currently available within Customer Call Reporter, but can also provide two additional new features:

- A scrolling message board, for important messages during the course of the working day
- A standings or “leader” board, which displays the top agents for a particular activity during the day

In addition to this, the wallboard can also be customized with the company’s logo and colors. Also, there is no limit to the number of statistics that can be displayed (caution: adding too many statistics may cause the displayed statistics to be too small to be viewed reasonably).

Wallboard Requirements

- The wallboard can run on all the supported browsers for Customer Call Reporter
- Microsoft Silverlight is a required plug in that can be downloaded before starting the wallboard

Wallboard Licensing

The Customer Call Reporter wallboard is based upon the number of supervisors licensed within CCR, for each supervisor license purchased; there can be one instance of the wallboard. If five supervisor licenses are active on the system, then five instances of the wallboard can be used throughout the business. The wallboard logins do not have to be unique, for example, the system may have only one wallboard account set up within CCR, but that same account can be used for as many supervisor licenses that exist on the system.

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Real Time Report/Monitor

Customer Call Reporter provides real time monitoring of call center activity. This is achieved via a thin client web browser session logged onto the IP Office Customer Call Reporter server. Within the supervisor client a user can program up to 3 different views of their call center. This is a very useful feature as now a supervisor can divide their responsibilities into different views and compare the results of those views.

For example, assume that a call center supervisor is responsible for the following hunt groups in IP Office; Computer Sales (201), Monitor Sales (202), Warranty Service (203), and non-Warranty service. The Supervisor can create two separate views entitled "sales" and "service" and compare the number of calls, length of call, etc. This allows a supervisor to see a whole system as it relates to the business function (sales, service).

The Supervisor views are private and can only be seen with a login. A view may be created that shows the system with all the agents, specific Hunt Group and Agent statistics as well as Alarms and Warnings. Each of the three views are summarized and made available to any Agents when they login and select their corresponding Supervisor. An example of the Supervisor real-time view is shown below:

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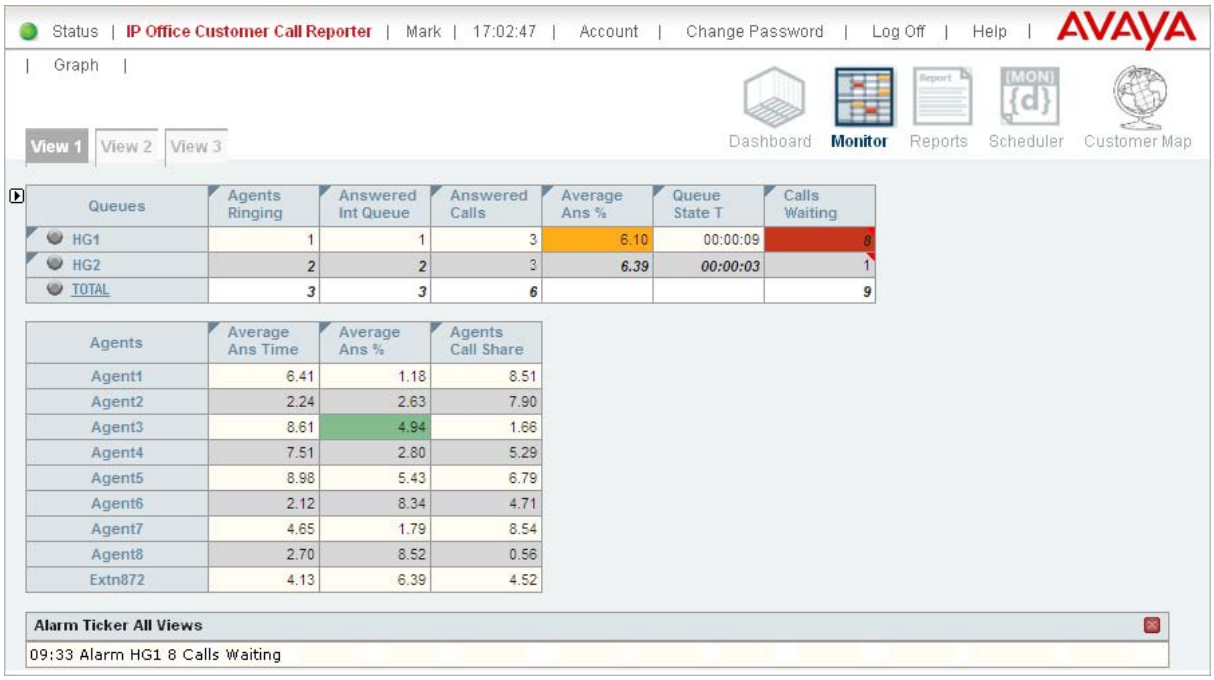
Status | IP Office Customer Call Reporter | Mark | 17:02:47 | Account | Change Password | Log Off | Help |
AVAYA
| Graph |
Dashboard
Monitor
Reports
Scheduler
Customer Map
View 1
View 2
View 3
Queues | Agents Ringing | Answered Int Queue | Answered Calls | Average Ans % | Queue State T | Calls Waiting
HG1 | 1 | 1 | 3 | 6.10 | 00:00:09 | 8
HG2 | 2 | 2 | 3 | 6.39 | 00:00:03 | 1
TOTAL | 3 | 3 | 6 | | | 9
Agents | Average Ans Time | Average Ans % | Agents Call Share
Agent1 | 6.41 | 1.18 | 8.51
Agent2 | 2.24 | 2.63 | 7.90
Agent3 | 8.61 | 4.94 | 1.66
Agent4 | 7.51 | 2.80 | 5.29
Agent5 | 8.98 | 5.43 | 6.79
Agent6 | 2.12 | 8.34 | 4.71
Agent7 | 4.65 | 1.79 | 8.54
Agent8 | 2.70 | 8.52 | 0.56
Extn872 | 4.13 | 6.39 | 4.52
Alarm Ticker All Views
09:33 Alarm HG1 8 Calls Waiting

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Alarms

The Customer Call Reporter provides different levels of alarms for the supervisor and agent to properly manage their call center activity. As parameters move from a normal to alarm state, the color of the field within the realtime view will change depending upon the level of alarm; white for normal, yellow for caution, red for alarm, and blue for alarm acknowledged. The following statistics can have alarms set against them within the IP Office Customer Call Reporter:

Statistic Full Name	Available for		Warning/ Alarm Type
	Queue	Agent	
Agent State (Queue) Time	-	✓	>1-999
Agents ACW	✓	-	>1-150
Agents Available	✓	-	<1-150
Agents Logged On	✓	-	<1-150
Answered Calls	✓	✓	>1-999
Answered External (Non-Queue)	-	✓	>1-999
Answered Internal (Non-Queue)	-	✓	>1-999
Answered Internal (Queue)	✓	✓	>1-999
Average Answer %	✓	✓	<1-100%
Average Answer Time	✓	✓	>1-600
Average Wait Time	✓	-	>1-600
Busy Not Available	✓	-	>1-150
Calls Waiting	✓	-	>1-999
Current Wait Time	✓	-	>1-600
Grade of Service	✓	-	<1-100%
Internal Made	-	✓	>1-999
Longest Wait Time	✓	-	>1-600
Lost Calls	✓	✓	>1-999
New Messages	✓	-	>1-999
No Answer	✓	✓	>1-999
Outbound Calls (External)	-	✓	>1-999
Overflowed Answered	✓	-	>1- 999
Overflowed Calls	✓	-	>1-999
Overflowed Calls Waiting	✓	-	>1-999
Overflowed Lost	✓	-	>1-999
Queue State Time	✓	-	>1-600
Routed to Other	✓	✓	>1-999
Routed to Voicemail	✓	✓	>1-999
Transferred	✓	✓	>1-999

Agent View

In Customer Call Reporter, the Agent view is part of the agent reporting license, every licensed agent will be able to run their own view. An Agent is able to login via a thin client web browser session and associate with a Supervisor. The Agent view also displays any alarms that have been set by the Supervisor.

Status | IP Office Customer Call Reporter | Agent3 | 17:02:47 | | Change Password | Log Off | Help | AVAYA
| Graph |
View 1 View 2 View 3
Queues | Agents Ringing | Answered Int Queue | Answered Calls | Average Ans % | Queue State T | Calls Waiting
HG1 | 1 | 1 | 3 | 6.10 | 00:00:09 | 8
HG2 | 2 | 2 | 3 | 6.39 | 00:00:03 | 1
TOTAL | 3 | 3 | 6 | | | 9
Agents | Average Ans Time | Average Ans % | Agents Call Share
Agent3 | 8.61 | 4.94 | 1.66

Call Statistics

The Customer Call Reporter provides detailed statistics to help small businesses understand exactly where a call has been within the business. Statistics are measured against (also refer to the table below):

- Queues
  Points in the call center where a queue forms such as hunt groups.

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- Agents
  Individuals that are members of queue groups who answer the calls.

Statistic Full Name	Available for ...	Call Type[1]		Warning/ Alarm Type	Include Overflow		Statistic Type
	Queue/Agent	Ext'	Int'		From	To	
Agent Productivity	✓/✓	✓	✗	>1-99%	✓	✗	Performance
Agent State (Queue)	–/✓	–	–	–	–	–	State
Agent State (Queue) Time	–/✓	–	–	>1-999[2]	–	–	State
Agent State (System)	–/✓	–	–	–	–	–	State
Agent State (System) Time	–/✓	–	–	–	–	–	State
Agents ACW	✓/–	–	–	>1-150	–	–	Current
Agents Available	✓/–	–	–	<1-150	–	–	Current
Agents Call Share	–/✓	–	–	–	–	✓	Performance
Agents Logged On	✓/–	–	–	<1-150	–	–	Current
Agents Present	✓/–	–	–	–	–	–	Current
Agents Ringing	✓/–	✓	✓	–	–	✗	Current
Answered Calls	✓/✓	✓	✓	>1-999	✗	✗/✓[3]	Call Count
Answered External (Non-Queue)	–/✓	✓	–	>1-999	–	–	Call Count
Answered Internal (Non-Queue)	–/✓	–	✓	>1-999	–	–	Call Count
Answered Internal (Queue)	✓/✓	–	✓	>1-999	✗	✗	Call Count
Average Answer %	✓/✓	✓	✓	<1-100%	✓	✗	Performance
Average Answer Time	✓/✓	✓	✓	>1-600	✓	✗	Performance
Average Wait Time	✓/–	✓	✓	>1-600	✓	✗	Performance
Busy Not Available	✓/–	–	–	>1-150	–	–	Current
Calls Waiting	✓/–	✓	✓	>1-999	✗	✓	Current
Current Wait Time	✓/–	✓	✓	>1-600	✓	✓	Performance
Grade of Service	✓/–	✓	✓	<1-100%	✓	✗	Performance
Internal Made	–/✓	–	✓	>1-999	–	–	Call Count
Longest Wait Time	✓/–	✓	✓	>1-600	✓	✗	Performance
Lost Calls	✓/✓	✓	✓	>1-999	✗	✗/✓[3]	Call Count
New Messages	✓/–	–	–	>1-999	–	–	Call Count
No Answer	✓/✓	✓	✓	>1-999	✗	✗/✓[3]	Call Count
Outbound Calls (External)	–/✓	✓	–	>1-999	–	–	Call Count
Overflowed Answered	✓/–	✓	✓	>1- 999	✓	✗	Call Count
Overflowed Calls	✓/–	✓	✓	>1-999	✓	✗	Call Count
Overflowed Calls Waiting	✓/–	✓	✓	>1-999	✓	✗	Call Count
Overflowed Lost	✓/–	✓	✓	>1-999	✓	✗	Call Count
Queue State	✓/–	–	–	–	–	–	State
Queue State Time	✓/–	–	–	>1-600[2]	–	–	State
Routed to Other	✓/✓	✓	✓	>1-999	✓	✗	Call Count
Routed to Voicemail	✓/✓	✓	✓	>1-999	✓	✗	Call Count
Talk Average	✓/✓	✓	✗	>1-600	✓	✗	Performance
Talk Inbound	✓/✓	✓	✗	–	✓	✗	Performance
Talk Inbound Average	✓/✓	✓	✗	–	✓	✗	Performance
Talk Internal	✓/✓	✗	✓	–	✓	✗	Performance
Talk Outbound	✓/✓	✓	✗	–	✓	✗	Performance
Talk Outbound Average	✓/✓	✓	✗	>1-600	✓	✗	Performance
Talk Total	✓/✓	✓	✗	–	✓	✗	Performance
Transferred	✓/✓	✓	✓	>1-999	✓	✗	Call Count

Customer Map

The customer map shows the location of callers based on the caller's number. When a caller's location has been identified , a pin is placed on the map at that location. The color of the pin changes with the volume of calls that match that same location. When the caller cannot be resolved beyond just a country, a colored button is used instead of a pin. Hovering your cursor over a pin or button displays details of the location and the number of callers from that location.

The customer map can use a number of different map backgrounds. These vary by map provider and map type.

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Map Type	Map Provider		
	Open Street Maps	Yahoo Maps	Outline Map
Road View			
Aerial View			

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Chapter 17. Management Utilities

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17. Management Utilities

This section gives an overview of the management applications that are common to all IP Office platforms.

[IP Office Manager]
IP Office's main configuration tool.

[IP Office Simplified Manager]
A flavor of the IP Office Manager targeted for Sub-20 type deployments.

[Monitor]
A trace utility for trouble shooting.

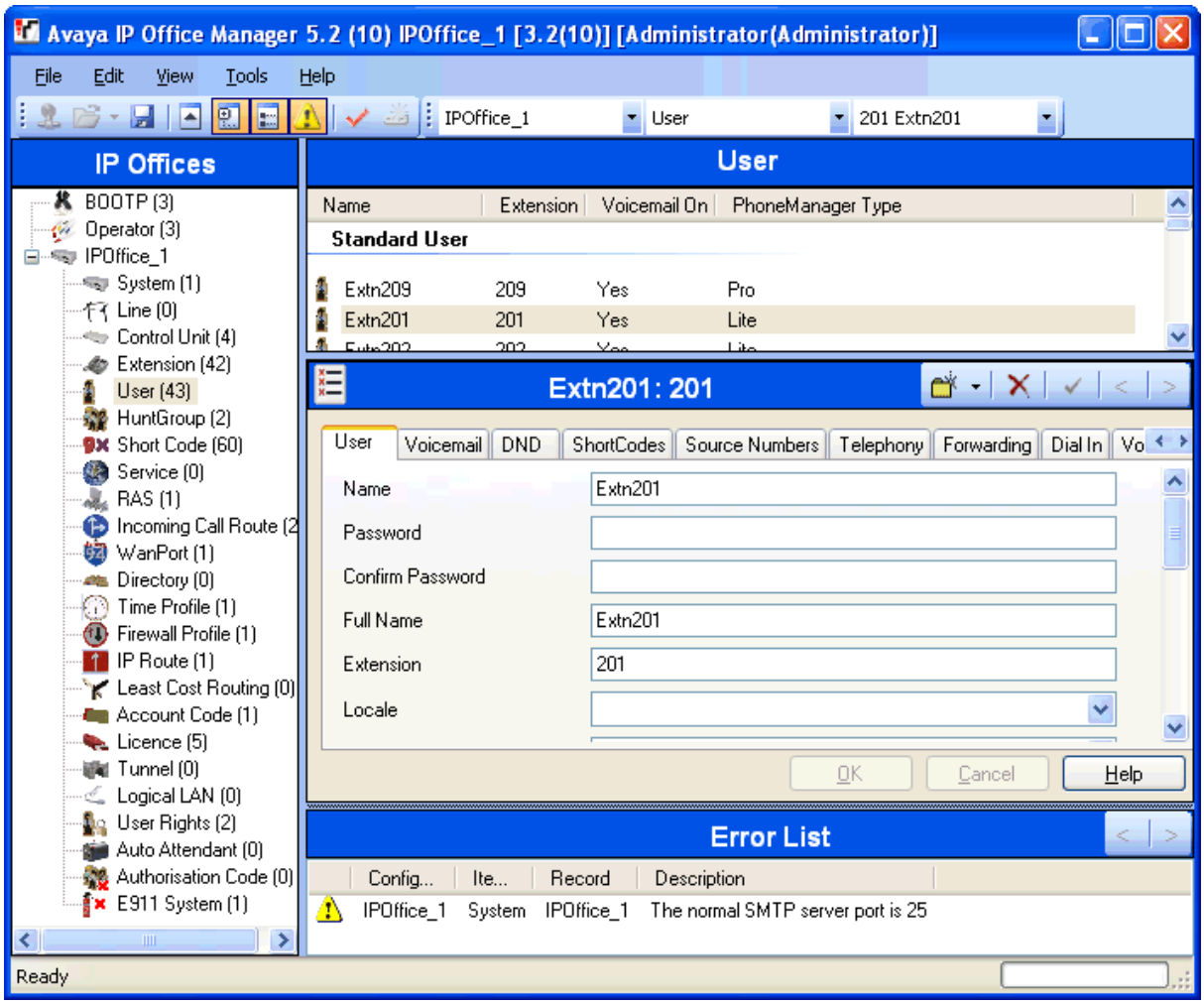
[SNMP]
Alerts and alarms from IP Office systems to SNMP tools or to SMTP email.

[System Status Application (SSA)]
A diagnostic tool to monitor and check the status of IP Office systems.

[Data Migration Manager (DMM)]
A migration tool to facilitate configuration data from BCM and Norstar® products to IP Office.

IP Office Manager

This application is IP Office's main configuration tool. Using a Windows Graphical User Interface, Manager provides an intuitive interface for installation, configuration and subsequent moves and changes. As with all IP Office applications, the Manager is multi-lingual and coupled with the ability to use the application both locally and remotely, it is possible for an administrator to manage any of their IP Offices from any country using their local language preference. Access to each IP Office is protected by passwords and definable user rights. This allows Manager to operate according to the individual administrator's level of expertise.



The IP Office Manager operates on a local copy of the IP Office configuration file. Configurations are prepared and reviewed 'off line' before committing to the IP Office. This has the benefit of ensuring a backup copy of the system configuration is always available for disaster recovery.

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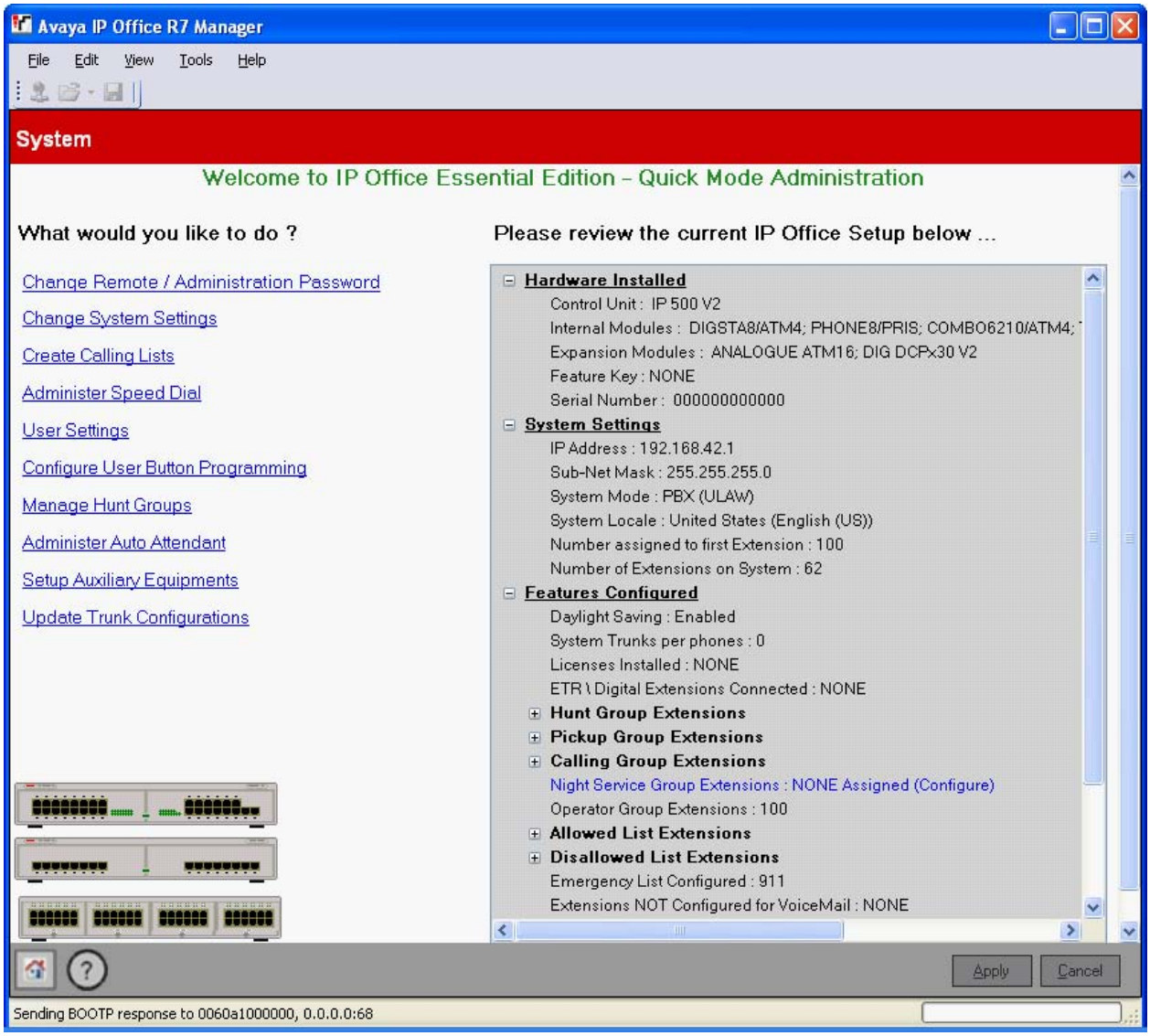
The IP Office Manager now supports IP Office Essential Edition Norstar Version and IP Office Essential Edition PARTNER® Version. In these versions, administrator can migrate from 2-digit dial plan to a 3-digit dial plan and vice-versa.

IP Office Simplified Manager for Quick Mode of Operation

IP Office Simplified Manager runs IP Office in Quick Mode. IP Office Essential Edition operates in two modes, namely Quick Mode and Standard Mode. Standard mode is the current mode of operation of IP Office and is managed using the current Windows GUI based thick client. The Quick Mode is managed by the Simplified Manager, a flavour of IP Office Manager designed to reduce the installation times, specifically for Sub-20 type deployments. This is achieved by enabling configuration of features important for this market segment. The Quick Mode further supports two sub-modes, namely Key Telephone System (KTS) Quick Mode and PBX Quick Mode. Based on the type of the SD card that is installed in the system, IP Office will boot up in one of the two Sub-Modes.

- µ-LAW SD Card: Default = Essential Edition KTS Quick Mode: North America and selected CALA countries
- A-LAW SD Card: Default = Essential Edition PBX Quick Mode: Rest of the World

The IP Office always boots up in Quick Mode by default. An option to migrate/restart manager in IP Office Standard Mode will be provided. When in Quick Mode the IP Office Manager offers task based configuration to speed up the installation process by providing easy and intuitive management.


Avaya IP Office R7 Manager
File Edit View Tools Help
System
Welcome to IP Office Essential Edition – Quick Mode Administration
What would you like to do ?
Change Remote / Administration Password
Change System Settings
Create Calling Lists
Administer Speed Dial
User Settings
Configure User Button Programming
Manage Hunt Groups
Administer Auto Attendant
Setup Auxiliary Equipments
Update Trunk Configurations
Please review the current IP Office Setup below ...
Hardware Installed
Control Unit : IP 500 V2
Internal Modules : DIGSTA8/ATM4; PHONE8/PRIS; COMBO6210/ATM4;
Expansion Modules : ANALOGUE ATM16; DIG DCPx30 V2
Feature Key : NONE
Serial Number : 000000000000
System Settings
IP Address : 192.168.42.1
Sub-Net Mask : 255.255.255.0
System Mode : PBX (ULAW)
System Locale : United States (English (US))
Number assigned to first Extension : 100
Number of Extensions on System : 62
Features Configured
Daylight Saving : Enabled
System Trunks per phones : 0
Licenses Installed : NONE
ETR \ Digital Extensions Connected : NONE
Hunt Group Extensions
Pickup Group Extensions
Calling Group Extensions
Night Service Group Extensions : NONE Assigned (Configure)
Operator Group Extensions : 100
Allowed List Extensions
Disallowed List Extensions
Emergency List Configured : 911
Extensions NOT Configured for VoiceMail : NONE
Apply
Cancel
Sending BOOTP response to 0060a1000000, 0.0.0.0:68

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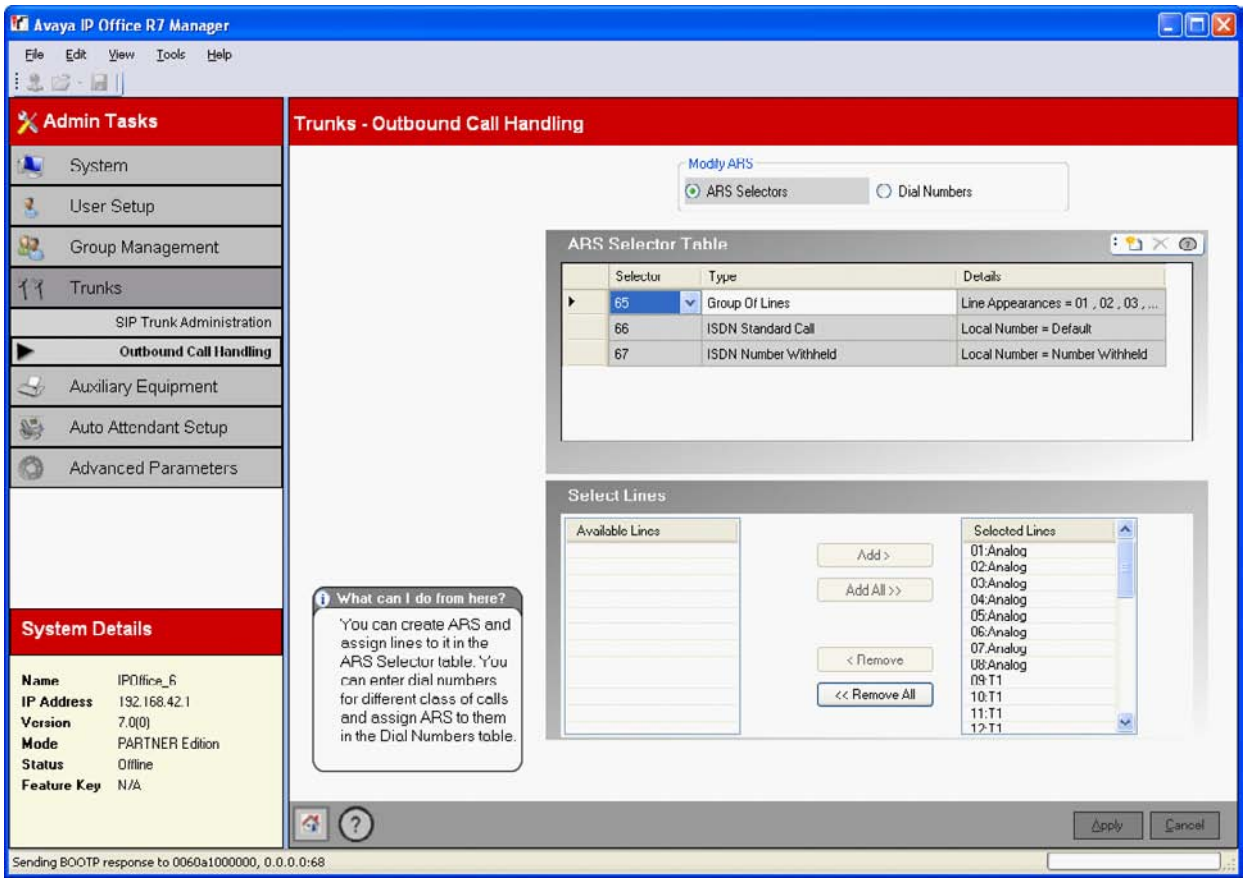

IP Office has a built-in audit trail that tracks changes to the system configuration, and who has made them. Manager can display the audit trail to assist with problem resolution. The audit trail records the last 15 changes in the configuration and records the following elements:

- Configuration Changed - For configuration changes, the log will report at a high level on all configuration categories (users, hunt group...) that have been changed.
- Configuration Erased
- Configuration merged
- Reboot – user instigated reboot
- Upgrade
- Cold Start
- Warm Start
- Write at HH:MM – This is when the administrator saved the configuration via the schedule option
- Write with Immediate Reboot
- Write with Reboot When Free

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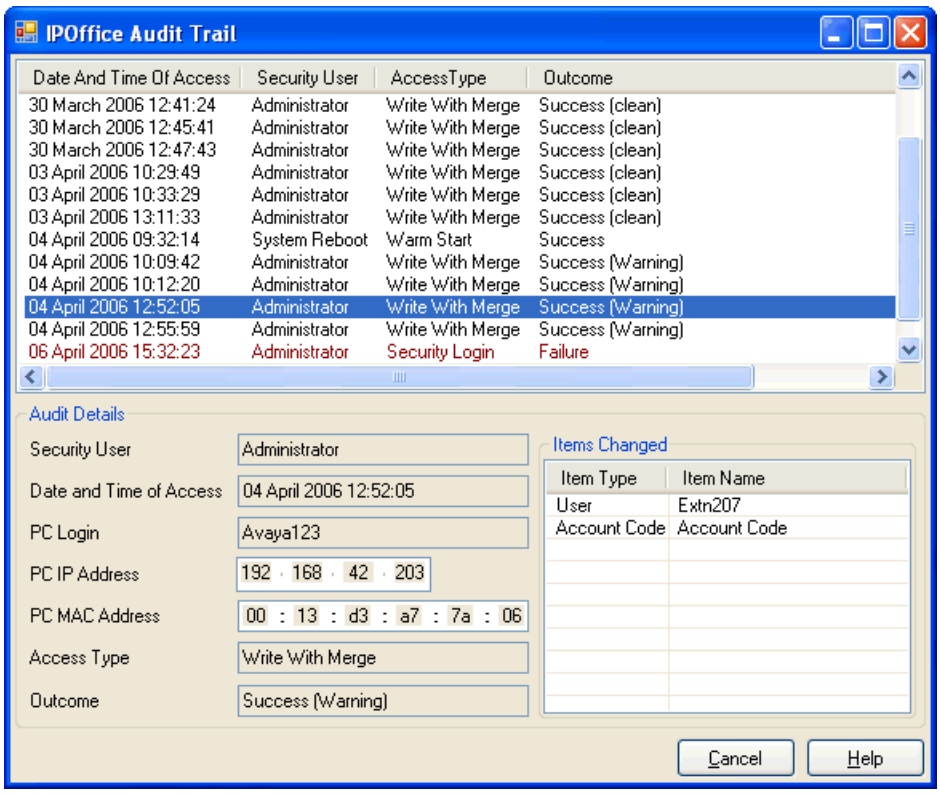
IPOffice Audit Trail
Date And Time Of Access
Security User
AccessType
Outcome
30 March 2006 12:41:24 Administrator Write With Merge Success (clean)
30 March 2006 12:45:41 Administrator Write With Merge Success (clean)
30 March 2006 12:47:43 Administrator Write With Merge Success (clean)
03 April 2006 10:29:49 Administrator Write With Merge Success (clean)
03 April 2006 10:33:29 Administrator Write With Merge Success (clean)
03 April 2006 13:11:33 Administrator Write With Merge Success (clean)
04 April 2006 09:32:14 System Reboot Warm Start Success
04 April 2006 10:09:42 Administrator Write With Merge Success (Warning)
04 April 2006 10:12:20 Administrator Write With Merge Success (Warning)
04 April 2006 12:52:05 Administrator Write With Merge Success (Warning)
04 April 2006 12:55:59 Administrator Write With Merge Success (Warning)
06 April 2006 15:32:23 Administrator Security Login Failure
Audit Details
Security User
Administrator
Date and Time of Access
04 April 2006 12:52:05
PC Login
Avaya123
PC IP Address
192 . 168 . 42 . 203
PC MAC Address
00 : 13 : d3 : a7 : 7a : 06
Access Type
Write With Merge
Outcome
Success (Warning)
Items Changed
Item Type
Item Name
User
Extn207
Account Code
Account Code
Cancel
Help

IP Office Manager is also used for maintenance functions such as:

- Upgrade to the IP Office system software
  - Systems running 2.1 or later have the added benefit of being able to send software over an IP network link to a system and have it validated before committing to the upgrade
- IP Office Manager Release is backwards compatible with systems from release 2.1 onwards to allow a single management application.
- Importing and Exporting IP Office configuration information in ACSII-CSV files. Manager will create files for the following data
  - Configuration.csv which is a complete list of items as per IP Office Manager and earlier
  - Directory.csv containing fields NAME, NUMBER
  - HuntGroup.csv containing fields HUNT GROUP NAME, HUNT GROUP EXTENSION, GROUP, HUNT, ROTARY, IDLE, QUEUING, VOICEMAIL, BROADCAST MESSAGES, EMAIL ADDRESS
  - License.csv is import only containing fields LICENCE OPTION, LICENCE KEY
  - ShortCode.csv containing fields SHORT CODE, TELEPHONE NUMBER, FEATURE NAME
  - User.csv containing fields NAME, EXTENSION NUMBER, USER RIGHT, EMAIL ADDRESS
- User templates for rapid programming and user rights for setting up user access levels

Monitor

The IP Office Monitor application is a real-time maintenance utility to assist with IP Office trouble-shooting. As the application connects to the IP Office over an IP connection it can be used from both local (LAN) and remote locations (WAN).

A simple interface allows an engineer to select which protocols and interfaces are to be monitored and decoded. The trace can either be captured directly to screen or as a log file for later analysis. Traces from different protocols can be color coded to improve the clarity of large log files. In addition to monitoring, the application captures system alarms and will display an activity log of the last 20 alarms that have occurred.

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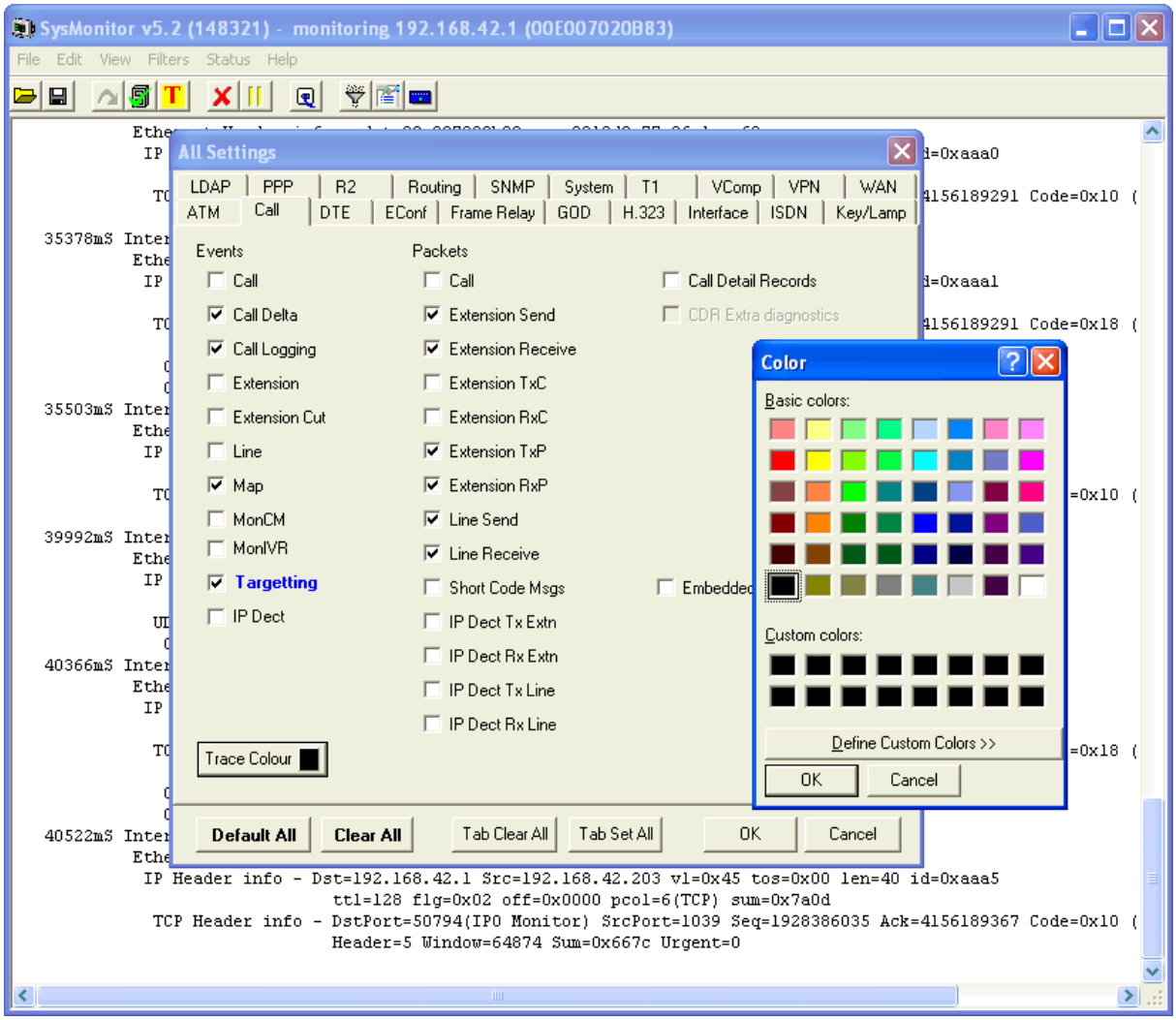

Simple Network Management Protocol (SNMP)

SNMP is an industry standard designed to allow the management of data equipment from different vendors using a single Network Manager application. The Network Manager will periodically poll equipment to solicit a response, if no response is received an alarm is raised. In addition to responding to polls, IP Office monitors the state of its Extensions, Trunk cards, Expansion Modules and Media cards so that if an error is detected IP Office will notify the Network Manager. IP Office allows two separate Network Managers to be configured so that both a customers Network Manager and a Maintainers Network Manager to be notified of the same alarm condition. As the IP Office solution comprises many applications, the core software notifies SNMP events from both VoiceMail Pro and Embedded voicemail to warn of approaching storage capacity limits.

IP Office has been tested against CastleRock's SNMPc-EE™ and HP's Network Node Manager (part of the OpenView application suite).

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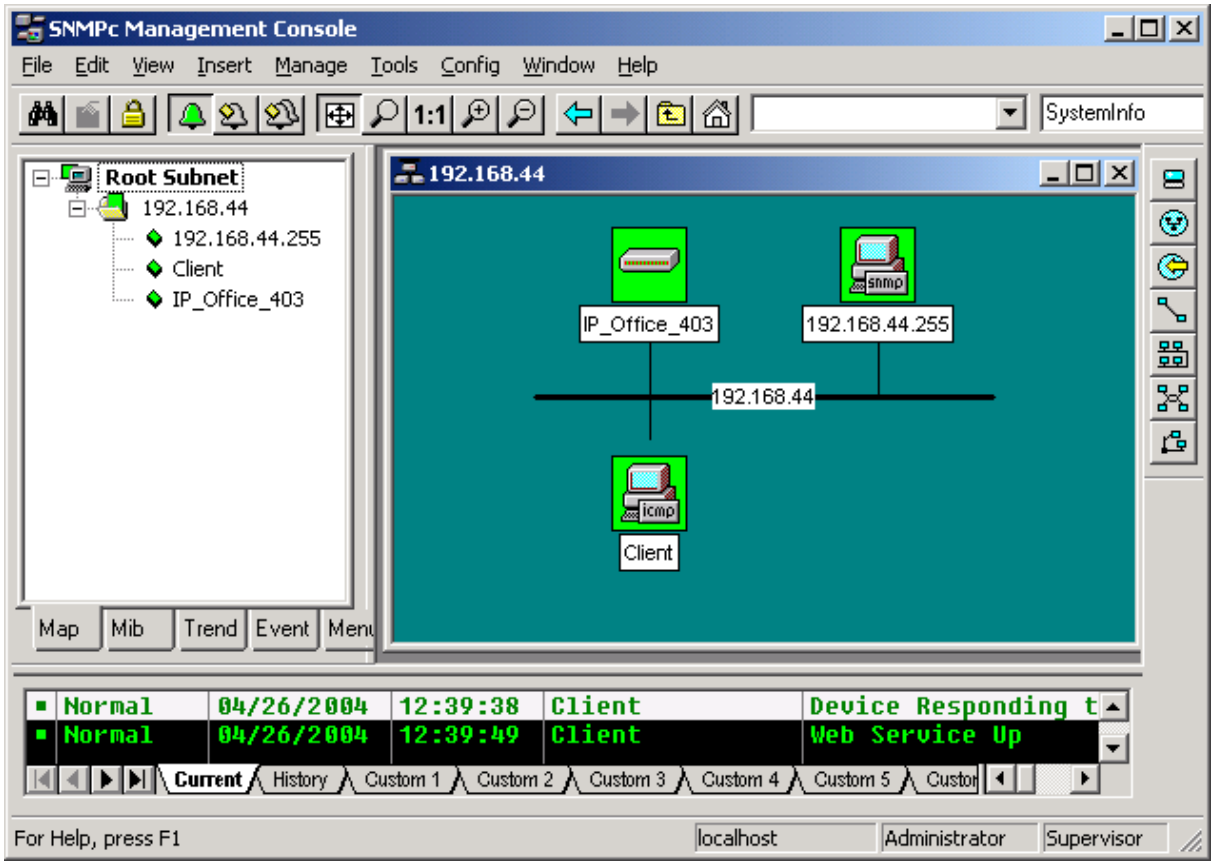

On customer sites where SNMP management is not available, IP Office can email events using up to 3 email addresses each containing a different set of alarms. The following system event categories can be chosen for email notification, if installed on the system:

- Generic
- Trunk lines
- Embedded Messaging Card
- VCM
- Expansion modules
- Applications
- License
- Phone change
- CSU Loop-Back

IP Office sends email notifications directly to the email server; no additional PC client is needed.

System Status Application

The System Status Application (SSA) is a diagnostic tool for system managers and administrators to monitor and check the status of IP Office systems locally or remotely. SSA shows both the current state of an IP Office system and details of any problems that have occurred. The information reported is a combination of real-time events, historical events, status and configuration data to assist fault finding and diagnosis. SSA provides real-time status, historic utilization and alarm information for ports, modules and expansion cards on the system. SSA connects to all variants of IP Office running release 4.0, using an IP connection that can be remote or local. Modem connections at 14.4kbps or above are supported for remote diagnostics.

SSA provides information on the following:

- Alarms
  SSA displays all alarms which are recorded within IP Office for each device in error. The number, date and time of the occurrence is recorded. The last 50 alarms are stored within IP Office to avoid need for local PC.
- Call Details
  Information on incoming and outgoing calls, including call length, call ID and routing information.

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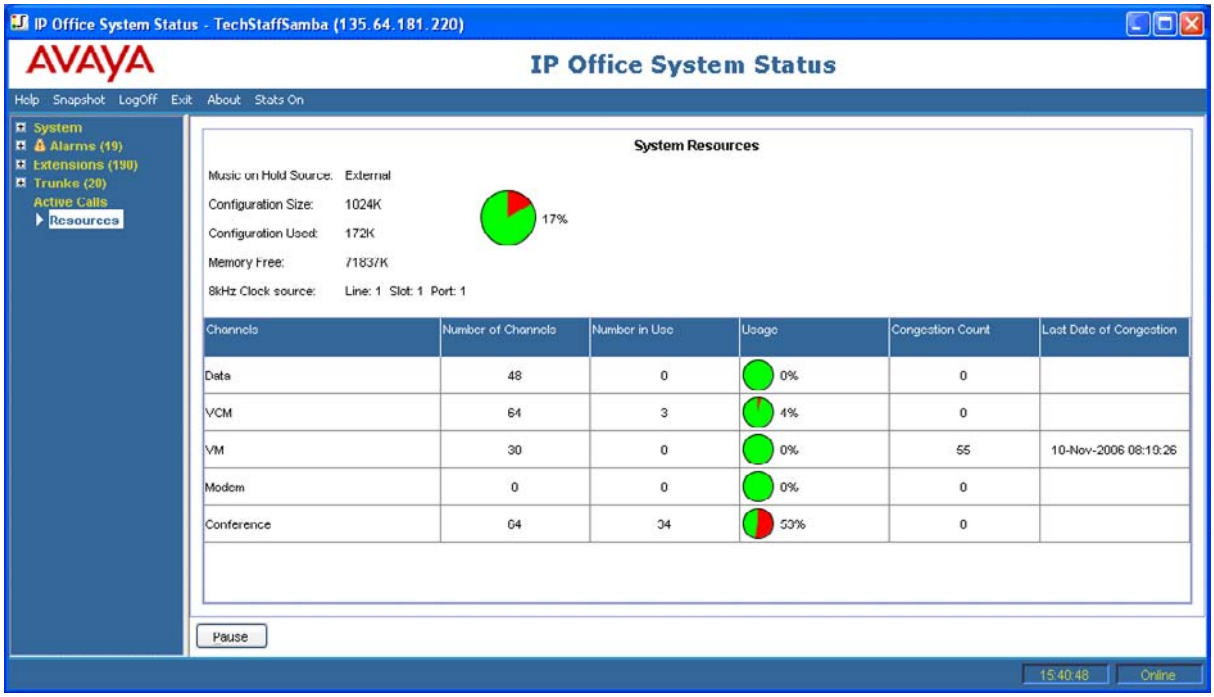
- Extensions
  SSA details all extensions (including device type and port location) on the IP Office system. Information on the current status of a device is also displayed.
- Trunks
  IP Office trunks and connections (VoIP, analog and digital) and their current status are displayed. For VoIP trunks, QoS information is also displayed (e.g. round trip delay, jitter and packet loss).
- System Resources
  IP Office includes central resources that are utilized to perform various functions. Diagnosing these resources is often critical to the successful operation of the system. This includes details on resources for VCM, Voicemail and conferencing.
- QoS Monitoring
  QoS Parameters from connected calls, such as jitter and roundtrip delay, are monitored.


IP Office System Status - TechStaffSamba (135.64.181.220)
AVAYA
IP Office System Status
Help Snapshot LogOff Exit About Stats On
System
Alarms (19)
Extensions (190)
Trunks (20)
Active Calls
Resources
System Resources
Music on Hold Source: External
Configuration Size: 1024K
Configuration Used: 172K
Memory Free: 71837K
8kHz Clock source: Line: 1 Slot: 1 Port: 1
17%
Channels | Number of Channels | Number in Use | Usage | Congestion Count | Last Date of Congestion
Data | 48 | 0 | 0% | 0 |
VCM | 64 | 3 | 4% | 0 |
VM | 30 | 0 | 0% | 55 | 10-Nov-2006 08:10:26
Modem | 0 | 0 | 0% | 0 |
Conference | 64 | 34 | 53% | 0 |
Pause
15:40:48
Online

SSA can be launched independently or from IP Office Manager and there can be up to two (2) SSA clients connected to an IP Office unit at one time.

Note: SSA is not a configuration tool for IP Office systems. For information on configuration, refer to IP Office Manager.

Data Migration Manager

The Data Migration Manager (DMM) is a migration tool designed to facilitate quick and easy migration from other products like BCM and Norstar to IP Office. The DMM tool is launched independently of the source and target applications. There are three main steps in the migration process.

1. Extract: This step includes specification of the source from which you intend to extract the configuration data file as well as other data including voicemails, greetings, recorded names etc.

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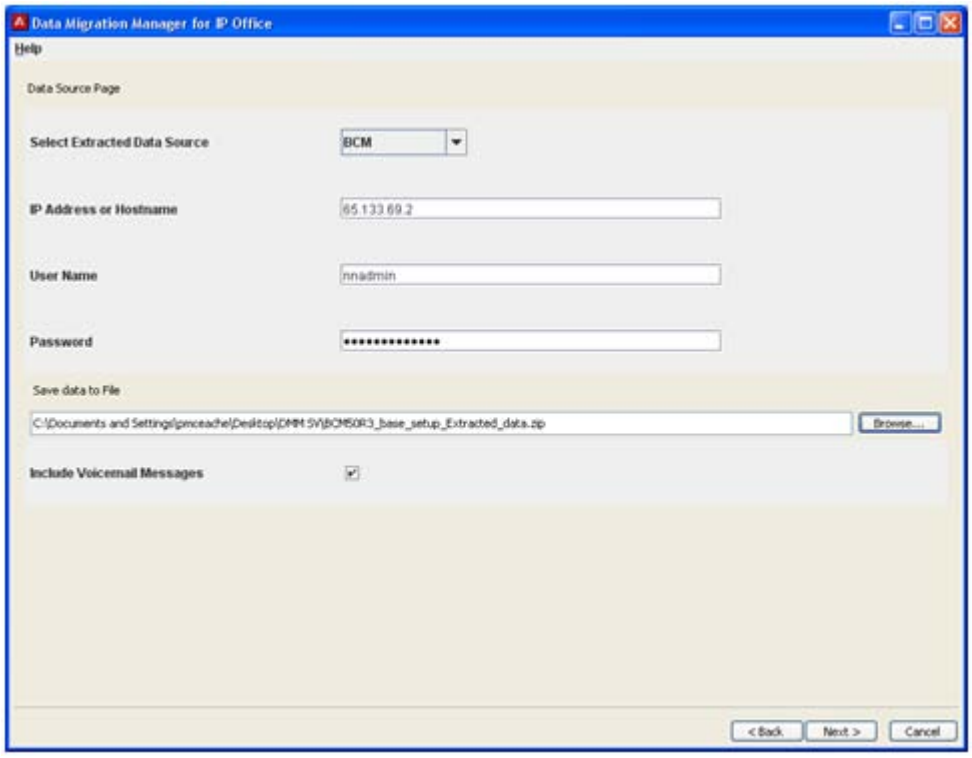
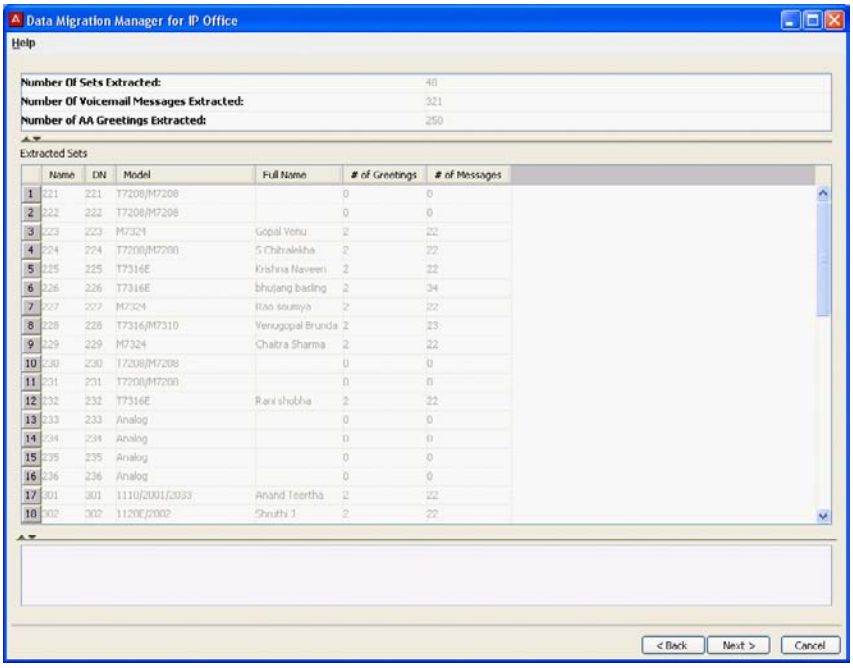


1. This step includes converting the data in the format of the source product to the format of the destination product. For example convert data that is understood by BCM to data that is understood by IP Office.



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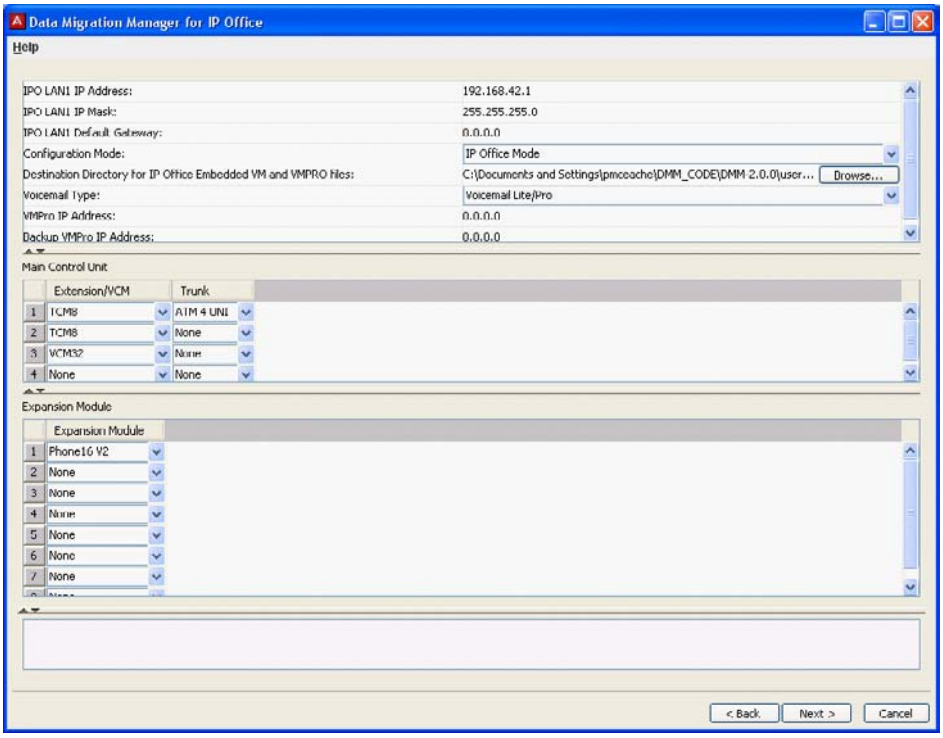
2. Apply:
This step includes applying the configuration to the destination product, which is always IP Office.



Data Migration Manager will eliminate the need to re-record announcements and greetings when transitioning from BCM or Norstar to IP Office. DMM will extract Nortel Call Pilot configuration and translate it into IP Office settings while moving greetings, announcements and voicemail messages off BCM/Norstar to IP Office. This will create a strong competitive advantage and reduce the installation times significantly.

BCM

- BCM200 4.0
- BCM400 4.0
- BCM1000 4.0
- BCM50 3.0, 5.0, 6.0
- BCM450 1.0, 5.0, 6.0

IP Office 7.0 and later

- Standard Mode
- Norstar Mode
- PARTNER Mode

IP500 V2 control unit required

Norstar

- CICS, MICS 7.x
- Call Pilot 100/150 3.1

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Chapter 18. IP Office Country Availability

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18. IP Office Country Availability

IP Office Country Availability

Country Availability

IP Office is available from distribution partners in the following countries. Please refer to your country price list for the availability of individual items.

- Argentina
- Brazil
- Canada
- Chile
- Mexico
- Peru
- USA
- Austria
- Bahrain
- Belgium
- Bulgaria
- Croatia
- Cyprus
- Czech Republic
- Denmark
- Estonia
- Egypt
- Finland
- France
- Germany
- Greece
- Hungary
- Iceland
- Ireland
- Israel
- Italy
- Kuwait
- Latvia
- Liechtenstein
- Lithuania
- Luxembourg
- Malta
- Morocco
- Netherlands
- Norway
- Poland
- Portugal
- Romania
- Russian Federation
- Saudi Arabia
- Slovak Republic
- Slovenia
- Slovakia
- South Africa
- Spain
- Sweden
- Switzerland
- Turkey
- UAE
- United Kingdom
- Australia
- China
- Hong Kong
- India
- New Zealand
- South Korea
- Taiwan
- Indonesia
- Macau
- Philippines
- Singapore
- Sri Lanka
- Thailand

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Chapter 19.
TAPI Functions Supported by IP Office

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19. TAPI Functions Supported by IP Office

TAPI 2.1 Functions Supported

TAPI*Link* Lite provides the following functionality for TAPI 2.1:

- lineAddToConference
- lineAnswer
- lineBlindtransfer
- lineCompleteTransfer
- lineConfigDialog
- lineClose
- lineDeallocateCall
- lineDial
- lineDrop
- lineGetAddressCaps
- lineGetAddressID
- lineGetAddressStatus
- lineGetAppPriority
- lineGetCallInfo
- lineGetCallStatus
- lineGetDevCaps
- lineGetID
- lineGetLineDevStatus
- lineHold
- lineInitialiseEx
- lineMakeCall
- lineNegotiateTAPIVersion
- lineOpen
- linePark
- lineRedirect
- lineRemoveFromConference
- lineSetAppPriority
- lineSetAppSpecific
- lineSetCallPrivilege
- lineSetStatusMessages
- lineSetupTransfer
- lineShutdown
- lineSwapHold
- lineUnhold
- lineUnpark
- lineSetCallData
- lineDevSpecific
- lineGenerateDigits
- lineGenerateTone
- lineMonitorDigits
- lineMonitorTones

TAPI 3.0 functions supported

The following functions are supported using TAPI 3.0:

- ITTAPI
- Initialize
- Shutdown
- EnumerateAddresses
- RegisterCallNotifications
- Put_EventFilter

- ITAddress
- get_AddressName
- get_dialableAddress
- get_ServiceProviderName
- CreateCall

- ITMediaSupport
- get_MediaTypes

- ITCallInfo
- get_Address
- get_CallState
- get_CallInfoString
- SetCallInfoBuffer

- ITBasicCallControl
- Connect
- Answer
- Disconnect
- Hold
- SwapHold
- ParkDirect
- Unpark
- BlindTransfer
- Transfer

- ITCallStateEvent
- get_Cause
- get_State
- get_Call

- ITCallNotificationEvent
- get_Call

- ITCallInfoChangeEvent
- get_Call

- ITCallHubEvent
- get_Event
- get_Call

Notes:

- TAPI*Link* Lite can be used from C, C++ and Delphi. Visual Basic cannot directly use TAPI 2.1, but does support TAPI 3.0 without any third-party tools.
- TAPI*Link* Lite provides detailed information on telephony events, including the ability to screen-pop based on CLI and/or DDI.

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Device-Specific Data

The following table shows the device specific data available via TAPI.

- Phone's extension number
- Forward on busy flag
- Forward on no answer flag
- Forward unconditional flag
- Forward hunt group flag
- Do not disturb flag
- Outgoing call bar flag
- Call waiting on flag
- Voicemail on flag
- Voicemail ring-back flag
- Number of voicemail messages
- Number of unread voicemail messages
- Outside call sequence number
- Inside call sequence number
- Ring back sequence number
- No answer timeout period
- Wrap up time period
- Can intrude flag
- Cannot be intruded upon flag
- X directory flag
- Force login flag
- Login code flag
- System phone flag
- Absent message id
- Absent message set flag
- Voicemail email mode
- User's extension number
- Users Locale
- Forward number
- Follow me number
- Absent text
- Do not disturb exception list
- Forward on busy number
- User's priority
- Number of groups the user is a member of
- Number of groups that the user is a member of that are currently outside their time profile
- Number of groups the user is currently disabled from
- Number of groups that the user is a member of that are currently out of service
- Number of groups that the user is a member of that are currently on night service

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DevLink Fields

The following table shows the device specific data available via DevLink.

#	Field Data (S Message)	#	Field Data (S Message)
1	A call id	26	Voicemail disallow
2	B call id	27	Sending complete
3	A state	28	Bc.tc,bc.tm
4	B state	29	Owner hunt group name
5	A connected	30	Original hunt group name
6	A is music	31	Original user name
7	B connected	32	Target hunt group name
8	B is music	33	Target user name
9	A name	34	Target RAS name
10	B name	35	Is internal call
11	B list (possible targets for the call)	36	Time stamp
12	A slot ,channel	37	Connected time
13	B slot , channel	38	Ring time
14	Called party presentation and type	39	Connected duration
15	Called party number	40	Ring duration
16	Calling party presentation and type	41	Locale
17	Calling party number	42	Park slot number
18	Called sub address	43	Call waiting
19	Calling sub address	44	Tag
20	Dialled party type	45	Transferring
21	Dialled party number	46	Sv active
22	Keypad type	47	Sv quota used
23	Keypad number	48	Sv quota time
24	Ring attempt count	49	Account code
25	Cause	50	Unique call identifier
#	Field Data (D Message)	#	Field Data (A Message)
1	A call id	1	A call id
2	B call id	2	B call id
3	Unique call identifier	3	Unique call identifier

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Chapter 20.
Technical Specifications

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20. Technical Specifications

PC requirements and specifiacations of the various IP Office applications can be found in a separate document in the IP Office section of support.avaya.com.

Dimensions

Unit Dimensions (mm/inches)	Width	Height	Depth
IP500 and IP500V2	445mmm/17.5"	73mm/2.9"	365mm/14.4"

- The recommended minimum clearance, front and rear, for the connection of cables and other devices is 75mm/3".

Weight

Unit	Weight
IP500 Control Unit	3.2Kg/7.0lbs
IP 500 V2 Control Unit	3.2Kg/7.0lbs
IP412 Control Unit	3.0Kg/6.7lbs
Analog 16 Module	2.9Kg/6.5lbs
DS16 Module	3.0Kg/6.7lbs
DS30 Module	3.5Kg/7.8lbs
DS16A Module	3.0Kg/6.7lbs
DS30A Module	3.5Kg/7.8lbs
TCM8	0.4Kg/0.9lbs
WAN3 Module	2.8Kg/6.3lbs
So8 Module	2.8Kg/6.3lbs
Phone 8 Module	2.8Kg/6.3lbs
Phone 16 Module	2.9Kg/6.5lbs
Phone 30 Module	3.1Kg/6.94lbs

Environmental

- 0°C to +40°C (32°F to 104°F). 95% relative humidity, non-condensing.

Call Capacity

Busy Hour Call Complete (BHCC) is a measure used to test IP Office systems under a high call load. For BHCC tests, each incoming call rings for 5 seconds, is answered and stays connected for 6 seconds. The BHCC figures for the different IP Office control units are listed below. Note that in some cases the BHCC figure achievable will be limited by the number of incoming trunks supported.

- IP500 = 36000.
- IP500 V2 = 36000.

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Telephone Extension Cable Lengths

The following table details the maximum cable lengths supported for the telephone ranges. These figures assume that standard twisted-pair telephone cable or CAT5 network cable is used.

Telephone	Unshielded Twisted-Pair (UTP) - 50nf/Km			CW1308
	AWG22 (0.65mm)	AWG24 (0.5mm)	AWG26 (0.4mm)	
1600/9600 Series	NA determined by network	NA determined by network	NA determined by network	NA determined by network
1400 Series	1200m/3937'	1000m/3280'	670m/2200'	400m/1310'
5400 Series	1200m/3937'	1000m/3280'	670m/2200'	400m/1310'
T3 Series (Upn)	1000m/3280'	1000m/3280'	400m/1310'	–
Analog telephones	1000m/3280'	1000m/ 3280'	400m/1640'	800m/2620'

Heat Dissipation

Note that the above numbers are for reference only. For practical purposes, for example the calculation of heat dissipation, it is recommended to base environmental requirements (for example air cooling or UPS ratings) on the maximum input rating of the power supplies of the planned IP Office configuration, as follows.

In order to calculate the maximum, that is worst case, amount of heat that can be generated by an IP Office system, it is assumed that all input power is converted to heat; whether from the PSU itself, the system unit, expansion module and/or cabling.

Heat dissipation is normally measured in British Thermal Units (BTU's). A heat value expressed in Watts can be converted to BTU/hr by multiplying by 3.41297. As indicated above, you should use the maximum power input of 115 VA of each power supply to calculate this most accurately.

Using the conversion factor:

- Heat Dissipation = 115 x 3.41297 = 392.5 BTU/hour.

The metric equivalent to BTU is a Joule where 1 BTU = 1,055 Joules.

This calculates the BTU value per power supply. The maximum BTU per system is therefore calculated, based on total number of power supplies installed in the system.

- IP500 V2 Maximum Heat Dissipation = 13 x 392.5 = 5,103 BTU/hr.

Remember to budget for the power requirements of any additional devices that are to be co-located with the IP Office such as server PC's (voicemail, etc).

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Power Supply

- Input
  - IP Office 500 V2 and IP Office 500 System Unit: IEC AC inlet socket. 100-240V AC, 50/60Hz, 81-115VA, 2.5A maximum.

- Power Supply Units: All CE/UL/Dentori Safety Approved.
  - Standard 40W Power Supply Unit (All control and expansion units unless otherwise indicated) Supplied with the control or expansion unit. 40W PSU with integral lead to the unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C7 power cord (2-wire figure 8 connector).
    - Input: 100-240V AC, 50/60Hz, 81-115VA, 2A maximum.
    - Output: 24Vdc, 1.875A, output power 45W maximum.
  - IP Office 500 80W internal Power Supply
    Integral to the System Unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C13 power cord (3-wire earthed cold kettle lead).
    - Input: 100-240V AC, 50/60Hz, 81-115VA, 2.5A maximum.

Interfaces

Interface	Information
DTE Port	• 9 way D-type D-Type female connector: V.24/V.28.
ISDN Ports	EU Interfaces: • BRI: RJ45 sockets. ETSI T-Bus Interface to CTR3 for Pan European Connection. • PRI E1: RJ45 socket. ETSI T-Bus Interface to CTR4 for Pan European Connection. • PRI T1/J1: RJ45 socket: FCC Part 68/JATE connection. USA Interfaces: • PRI T1 Service: Ground Start (GS) – Default, E&M, 56k data for 5ESS, 56/64/64 restricted for 4ESS. • PRI ISDN Switch support: 4ESS, 5ESS, DMS-100, DMS-250 (includes conformance to ANSI T1.607 and Bellcore Special Report SR4287, 1992). • PRI ISDN Services: AT&T Megacom 800, AT&T WATS (4ESS), AT&T SDS Accunet 56kB/s and 64kB/s (4ESS), AT&T Multiquest (4ESS).
Analog Trunk Ports	• RJ45 sockets: Loop start/Ground start (regional dependant)
Power Fail Ports	• RJ45 sockets:
ISDN Data Rates	• BRI: B-channel 64kbps or 56kbps, D-channel 16kbps. • PRI: B-channel 64kbps or 56kbps, D-channel 64kbps.
Analog Phone Ports	• RJ45 sockets • REN: 2. (External Bell via POT port: REN = 1) • Off Hook Current: 25mA. • Ring Voltage: 40V (nominal) RMS.
LAN	• RJ45 sockets. Auto-negotiating 10/100 BaseT Ethernet (10/100Mbps).
WAN	• IP406 V2 and IP412: 37 way D-Type female sockets. X.21 interface to 2048k bps, V.35 interface to 2048Kbps and V.24 Interface to 19.2Kbps.
Audio	• 3.5mm Stereo Jack socket. Input impedance - 10k /channel. • Maximum AC signal – 200mV rms.
External Output Port	• 3.5mm Stereo Jack socket. Switching Capacity - 0.7A. • Maximum Voltage - 55V DC. On state resistance - 0.7. • Short circuit current - 1A. Reverse circuit current capacity - 1.4A.
Embedded Voice Memory	• IP406 V2 and IP500: 512MB Compact Flash memory card.

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Protocol	RFC	Information
V120	-	A standard Rate Adaptation mechanism.
V110	-	A standard Rate Adaptation mechanism.
PPP	RFC1661	Point to Point Protocol.
LCP	RFC1570	Link Control Protocol.
MP	RFC1990	Multi-Link (Point to Point) Protocol.
IPCP	RFC1332	Internet Protocol Control Protocol.
PAP	RFC1334	Password Authentication Protocol.
RTP/RTCP	RFC1889	Real Time and Real Time Control Protocol.
CHAP	RFC1994	Challenge Handshake Authentication Protocol.
CCP	RFC1962	Compression Control Protocol.
STAC	RFC1974	STAC LZS Compression Protocol.
MPPC	RFC2118	Microsoft Point to Point Compression (Protocol).
BACP	RFC2125	Bandwidth Allocation Control Protocol.
UDP	RFC768	User Datagram Protocol.
IP	RFC791	Internet Protocol.
TCP	RFC793	Transmission Control Protocol.
DHCP	RFC1533	Dynamic Host Control Protocol.
NAT	RFC1631	Network Address Translation.
BOOTP	RFC951	Bootstrap Protocol.
TFTP	RFC1350	Trivial File Transfer Protocol.
NTP	RFC868	Network Time Protocol.
SNMPv1	RFC1157	Simple Network Management Protocol. (STD15)
	RFC1155	Structure and identification of management information for TCP/IP based internets. (STD16)
	RFC1212	Concise MIB Definitions. (STD16)
	RFC1215	A convention for defining traps for use with SNMP.
MIB-II	RFC1213	Managment Information base for network management of TCP/IP based internets: MIB-II. (STD17)
ENTITY MIB	RFC2737	Entity MIB (Version 2).
RIP	RFC1058	Routing Information Protocol.
	RFC2453	RIP Version 2. (STD56)
	RFC1722	RIP Version 2 Protocol Applicability Statement. (STD57)
IPSec	RFC2401	Security Architecture for the Internet Protocol.
	RFC2402	IP Authentication Header.
	RFC2403	The Use of HMAC-MD5-96 within ESP and AH.
	RFC2404	The Use of HMAC-SHA-1-96 within ESP and AH.
	RFC2405	The ESP DES-CBC Cipher Algorithm with Explicit IV.
	RFC2406	IP Encapsulation Security Payload. (ESP)
	RFC2407	The Internet IP Security Domain of Interpolation for ISAKMP.
	RFC2408	Internet Security Association and Key Management Protocol.
	RFC2409	The Internet Key Exchange.
	RFC2410	The NULL Encryption Algorithm and its Use with IPSec.
	RFC2411	IP Security Document Roadmap.
L2TP	RFC2661	Layer Two Tunneling Protocol "L2TP".
	RFC3193	Securing L2TP using IPSec.
Header Compression	RFC2507	IP Header Compression (IPHC).
	RFC2508	Compressing IP/UDP/RTP Headers for Low-Speed Serial Links.
	RFC2509	IP Header Compression over PPP.
DiffServ	RFC2474	Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers.
PPP MP	RFC1990	The PPP Multilink Protocol (MP).
Frame Relay Encapsulation	RFC1490	Multi protocol Interconnect over Frame Relay.

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Protocol	RFC	Information
ML-PPP	RFC2686	The Multi-Class Extension to Multi-Link PPP.

Session Initiation Protocol

- Rec. E.164 [2] - ITU-T Recommendation E.164: The international public telecommunication numbering plan
- RFC 2833 [7] - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3261 [8] - SIP: Session Initiation Protocol
- RFC 3263 [10] - Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264 [11] - An Offer/Answer Model with Session Description Protocol (SDP)
- RFC 3323 [14] - A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3489 [18] - STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
- RFC 3824 [24] - Using E.164 numbers with the Session Initiation Protocol (SIP)
- RFC 1889 – RTP
- RFC 1890 - RTP Audio
- RFC 4566 – SDP
- RFC 3265 - Event Notification
- RFC 3515 - SIP Refer
- RFC 3842 - Message Waiting
- RFC 3310 – Authentification
- RFC 2976 – INFO
- RFC 3323 - Privacy for SIP (PAI) and draft-ietf-sip-privacy-04 (RPID)

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Chapter 21. Licenses

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21. Licenses

Many solutions, applications and features on IP Office systems are licensed and only operate when a valid license is detected. This includes features within IP Office applications running on PCs connected to the IP Office system.

Licenses are 32 character strings derived from a unique serial number for the IP Office system which will be using the license. The unique serial number is taken from the Feature Key device fitted to the IP Office system.

- IP500 Smart Card Feature Key
  For IP500 IP Office control units, the serial number is based on the Smart Card Feature Key fitted to the control unit. A Smart Card Feature key is mandatory on all IP500 control units for correct operation of the unit regardless of whether the unit is using any licenses.
- IP500 V2 SD Card Feature Key
  For IP500 V2 control units, the serial number is based on the System SD Card Feature Key fitted to the control unit. A System SD Card Feature key is mandatory on all IP500 V2 control units for correct operation of the unit regardless of whether the unit is using any licenses.

  There are three types of System SD card available, which define the default operating mode:

  - System SD CARD A-LAW
  - System SD CARD MU-LAW
  - System SD CARD PARTNER
  - System SD CARD NORSTAR

For licensing each IP Office control unit must have its own Feature Key and licenses. The Feature Key associated with one IP Office system cannot be used to validate the licenses of another IP Office system.

Since licenses are based on the serial number of the Feature Key, they cannot be moved between systems with different Feature Keys. If a Feature Key is lost the license will become invalid. If it is necessary to exchange a Feature Key due damage or upgrade then the existing key and licenses can be exchanged (license swap) through the Avaya Direct International (ADI) website (https://adi.avaya.com).

Some licenses enable a number of ports, channels or users. Depending on the particular license, each supported instance may be consumed either when it is configured for use or alternatively when it is actually being used. When no further instances of a license are available, further use of the licensed feature is not allowed.

Typically for port, channel and user licenses, multiple licenses can be added to give a higher number of supported ports, channels or users. However licenses above the capacity supported by the particular IP Office control unit will not work.

The following sections detail the current individual IP Office licenses. The actual licenses are indicated by a icon and the following information:

- License Name in IP Office Configuration: *Avaya orderable part name (Material Code)*
  License description.

IP Office Upgrade License

From Release 6 onwards IP Office needs a valid Release Entitlement to run. For new IP500 V2 systems the Release Entitlement will be free of charge delivered with the SD Card. To upgrade an older IP Office system to Release 7.0 an Upgrade license is required.

- Upgrading to Release 7.0 – up to 32 users: *IPO LIC UPG R7.0 SML (262645)*
  Single control unit with no external expansion modules.
- Upgrading to Release 7.0 – more than 32 users: *IPO LIC UPG R7.0 (262644)*

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IP Office Group Collaboration Licenses

IP Office Preferred Edition

The following license is supported on all current IP Office control units.

- Preferred Edition: *IPO LIC PREFRD (VM PRO) RFA LIC:DS (171991)*
  In particular, this license enables:
    - advanced messaging
    - multi-level automated attendant
    - secure meet-me conferencing
    - call recording
    - conditional call routing
    - Text to Speech (TTS) for e-mail reading for users with appropriate User profiles
    - queue announcements

  This system license enables VoiceMail Pro and provides the initial 4 ports.

  IP Office Preferred Edition Messaging (VoiceMail Pro) is a messaging server application that runs on a Windows server PC. It is enabled through the Preferred Edition license (see above). Messaging services to multiple IP Offices can be supported when they are connected in an IP Office Small Community Network (SCN). In this centralized voicemail scenario, only the central IP Office requires the Preferred Edition license. The other IP Offices within the network do not require a Preferred Edition license. However, from Release 6 onwards due to load balancing, capacity increase and resilience reasons it is possible to have more than one Preferred Edition License in the SCN.

  By default Preferred Edition provides standard voicemail mailbox functions to all IP Office users and hunt groups plus support for call recording and call waiting announcements to callers. A range of additional features can be enabled by further licenses.

  - Additional Messaging Ports Licenses
    The Preferred Edition license includes support for 4 messaging ports. The total number of messaging ports supported can be increased by adding combinations of the following licenses, at the time of purchase or later, up to the maximum supported by the particular IP Office control unit (IP500/IP500 V2 = 40).
      - Additional Messaging (2 Ports): *IPO LIC VM PRO RFA 2 LIC:CU (174459)*
      - Additional Messaging (4 Ports): *IPO LIC VM PRO RFA 4 LIC:CU (174460)*
      - Additional Messaging (8 Ports): *IPO LIC VM PRO RFA 8 LIC:CU (174461)*
      - Additional Messaging Pro (16 Ports): *IPO LIC VM PRO RFA 16 LIC:CU (174462)*
  - Messaging Feature Licenses
    The Advanced Edition license provides Customer Call Reporter, ContactStore Call Recording Library, Interactive Voice Response (IVR), Visual Basic scripting and generic Text-To-Speech (TTS). The following licenses enable specific Messaging features beyond those already provided with Advanced Edition. These additional licenses all require the Preferred Edition (Voicemail Pro) license as a pre-requisite.
      - UMS Web Services Licenses (also enabled by the “Teleworker” and "Power User" licenses)
        UMS (Unified Messaging Server) allows users to access the messages in their Voicemail Pro mailbox via either a web browser (requires IIS on the Voicemail Pro server), using a IMAP compatible email application or using Microsoft Exchange Server 2007 message store. The total number of users supported is set by adding combinations of the following licenses. Note that a license is consumed for each user configured for UMS access.
          - UMS Web Services - 1 User License: *IPO LIC VMPRO UMS 1 USER (217880)*
      - Messaging Recordings Administrators License: *IPO LIC CONTACTSTORE RFA LIC:DS (187166)* (also included in the "Advanced Edition" license)
        As standard, Preferred Edition supports automatic and or manual call recording to specified mailboxes. Also provided as part of Advanced Edition, this license allows Preferred Edition to use a 3rd-party application to support the storage and administration of call recordings. Currently the supported application for this is ContactStore for IP Office. When used manual and or automatic records calls can be routed (along with the call details) to the ContactStore applications database for storage and retrieval when required.
      - Text to Speech (TTS) Licenses
        TTS can be used within customized voicemail call flows to speak information to callers rather than having to record prompts for the call flow or speak results of IVR queries. Two options are supported for TTS operation:
          - As part of the Advanced Edition, that enables the use of the default Text-To-Speech (TTS) speech engine supplied as part of Windows operating system or any other SAPI compliant speech engine. This allows the creation of automated call flows that speak results of IVR queries. Eight (8) licenses are provided with the Advanced Edition system license.

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- Messaging TTS (ScanSoft) License: *IPO LIC AVAYA TTS RFA 1 LIC:CU (182299)*
  This license enables the use of the TTS speech engines supplied as part of the IP Office Voicemail Pro software set. Multiple licenses can combined for the number of simultaneous uses of TTS required up to the number of voicemail channels licensed.
- Networked Messaging License: *IPO LIC NTWKD MSGING RFA LIC:DS (182297)*
  Voicemail Pro Networked Messaging (VPNM) can be used between separate IP Office systems, each with their own Voicemail Pro server, to exchange messages left for different mailboxes. This option is not supported within a Small Community Network (SCN) but can be used between separate SCN networks.

IP Office Advanced Edition (Release 6 onwards)

- Advanced Edition: *IPO LIC R6 ADV EDITION RFA LIC:DS (229424)*
  In particular, this license enables:
  - IP Office Customer Call Reporting (CCR) including 1 supervisor.
  - ContactStore Call Recording Library
  - Interactive Voice Response (IVR) with external databases
  - Visual Basic Scripting
  - 8 ports of Text-to-Speech (TTS) to allow database queries using IVR to be read over the phone.
  - The Advanced Edition needs a valid Preferred Edition license as a pre-requisite to run.

Essential Edition Additional Voicemail Ports License (IP500 V2 only)

Essential Edition provides embedded messaging does not require the presence of a license. However, from Release 6 onwards each IP500 V2 system supports Essential Edition by default with 2 ports of voicemail enabled. This can be expanded up to 6 channels through licenses in steps of two.

- Additional Embedded Messaging (2 Ports): *IPO LIC R6 ESSNTL EDITION ADD 2 (229423)*

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IP Office User Profile Licenses

The following chart shows an overview of the functions of each User Profile. For users in an Small Community Network (SCN), when they hot desk to another IP Office system, they retain their licensed profile setting as configured on their home system with the exception of Receptionist.

User Profile	Functionality Enabled								
	Mailbox	Mobility	Avaya one-X™ Portal for IP Office	Softphone	Unified Messaging	Text-to-Speech (Generic) for email reading	Soft Console	CCR Agent	CCR Supervisor
Basic User	✓								
Office Worker	✓		✓		✓				
Teleworker	✓		✓	✓	✓				
Mobile User	✓	✓				✓			
Power User	✓	✓	✓	✓	✓	✓			
Add ons									
Receptionist									
Customer Service Agent								✓	
Customer Service Supervisor									✓

Basic User

This functionality would be suitable for common areas (e.g. meeting rooms) or users with less communication needs. A basic user has all basic core functions enabled, including mailbox and VPN phone usage. This is the default functionality for users on the system when no licenses are applied.

Mobile Worker

This User Profile is targeted at users with mobile devices and no internet connection to the office, e.g. field sales and service staff who are often on the road. They are provided with functions such as one number access and call control as if "in the office", enabling them to never miss a call.

The Mobile Worker license enables a user to be configured to use the IP Office Mobility features. Those features include Mobile Twinning, Mobile Call Control (IP500 only), one-X Mobile Client and use of Text to Speech (TTS) for listening to your emails remotely. The license instances are consumed by a user being configured for any mobility feature.

Release 6 onwards

- Mobile Worker License - 1 User: *IPO LIC R6+ MOBILE WORKER 1 (229434)*
- Mobile Worker License - 5 Users: *IPO LIC R6+ MOBILE WORKER 5 (229435)*
- Mobile Worker License - 20 Users: *IPO LIC R6+ MOBILE WORKER 20 (229436)*

Office Worker

This User Profile is targeted at users who are working in the office. In conjunction with a wired or wireless Avaya Phone, they are provided with functions to control their communication with a PC e.g. control audio conferences or enabling them to control the voicemails with an e-mail client (this requires Preferred Edition).

The Office Worker license enables a user to use Unified Messaging Service (UMS) and one-X Portal for IP Office (without Telecommuter functionality). The license instances are consumed for each user being configured.

- Office Worker License - 1 User: *IPO LIC R6 OFF WORKER RFA 1 (229438)*
- Office Worker License - 5 Users: *IPO LIC R6 OFF WORKER RFA 5 (229439)*
- Office Worker License - 20 Users: *IPO LIC R6 OFF WORKER RFA 20 (229440)*

Teleworker

This User Profile is targeted at users who are working from home (or elsewhere) with Internet connection to the office. They are provided with the ability to leverage the PSTN/ISDN via Telecommuter mode, without incurring any remote phone charges. Alternatively the new IP Office Softphone can be used if the quality of service of the local internet connection is adequate.

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The Teleworker license enables a user to use one-X Portal for IP Office and the IP Office Video Softphone. UMS functionality for voicemail/email integration is also enabled, providing synchronization with any IMAP e-mail client (this requires Preferred Edition). The license instances are consumed for each user being configured.

Release 6 onwards

- Teleworker License - 1 User: *IPO LIC R6 TELEWORKER 1 (229430)*
- Teleworker License - 5 Users: *IPO LIC R6 TELEWORKER 5 (229431)*
- Teleworker License - 20 Users: *IPO LIC R6 TELEWORKER 20 (229432)*

Power User

This User Profile is targeted at key knowledge workers with the need for both Mobile Worker and Teleworker functionality.

The Power User license enables a user to use one-X Portal for IP Office, Softphone, all Mobility features and Unified Messaging Service (UMS). The license instances are consumed for each user being configured.

Release 6 onwards

- Power User License - 1 User: *IPO LIC R6 PWR USER 1 (229426)*
- Power User License - 5 Users: *IPO LIC R6 PWR USER 5 (229427)*
- Power User License - 20 Users: *IPO LIC R6 PWR USER 20 (229428)*

Receptionist

This User profile is targeted at operators/receptionists and provides a visual PC interface for call handling and management for multiple sites. Up to 4 of these users are supported on each site.

The Receptionist license enables a user to use IP Office SoftConsole. This is a Windows PC application intended for receptionists and operators. Up to 4 SoftConsoles can be licensed on a single IP Office system.

- Receptionist User License: *IPO LIC RECEPTIONIST RFA LIC:CU (171987)*
  Enables the 1st instance and subsequent instances of the PC-based operator console. Additional licenses can be added for up to 4 SoftConsoles running at any time.

Customer Service Agent

This User Profile is targeted at employees who are responsible for fielding customer service calls. This agent license enables a user to use IP Office Customer Call Reporter. It provides them with a web browser interface to view key statistics in real-time and also allows supervisor to track their performance with IP Office Customer Call Reporter. Up to 150 agents can be licensed on IP Office Customer Call Reporter.

- Customer Service Agent License - 1 User: *IPO LIC CUSTMR SVC AGT RFA 1 (217650)*
- Customer Service Agent License - 5 Users: *IPO LIC CUSTMR SVC AGT RFA 5 (217651)*
- Customer Service Agent License - 20 Users: *IPO LIC CUSTMR SVC AGT RFA 20 (217653)*

Customer Service Supervisor

This User Profile enables service supervisors to track, measure and create reports for agent (or a group of agents) productivity via a web browser interface. This supervisor license enables a supervisor to use IP Office Customer Call Reporter. Up to 30 supervisors can be licensed on IP Office Customer Call Reporter.

- Customer Service Supervisor License - 1 User: *IPO LIC R6 CUSTMR SVC SPV 1 (229442) Release 6 onwards*

CCC to CCR Migration

This upgrade license will convert ALL Compact Contact Center (CCC) licenses to Customer Call Reporter (CCR). This includes all supervisors and agents.

- CCC to CCR Upgrade License: *IPO LIC CUSTMR CUSTMR CALL REPORTER UPG LIC (217658)*

Phone Manager Licenses

IP Office Phone Manager is a legacy Windows PC application that allows IP Office users to control their telephone through their PC. An optional IP Softphone is also available with additional licensing. In Release 6, it is recommended to use one-X Portal for IP Office and the new IP Office Softphone (using Teleworker or Power User licenses) instead of Phone Manager Pro.

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- Phone Manager Pro License
  These licenses control how many users configured for Phone Manager Pro can simultaneously run Phone Manager in that mode. Multiple licenses can be combined.
  - Phone Manager Pro (per seat) - 1 User: *IPO LIC PMGR PRO RFA 1 LIC:CU (177468)*
- Phone Manager Softphone Licenses
  These licenses control how many users configured for Phone Manager Pro PC Softphone can simultaneously run Phone Manager in that mode. Note that running Phone Manager in this mode also requires and consumes a Phone Manager Pro license for each user.
  - Phone Manager Softphone (per seat) - 1 User: *IPO LIC PMGR SFTPHN RFA 1 LIC:CU (171992)*

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IP Office 500 System Licenses

The following licenses are used to enable features for systems based on the IP500 and IP500 V2 control units.

- IP500 Universal PRI Additional Channel Licenses
  The IP500 PRI Universal trunk card (IP500 PRI-U) is available in 1 or 2 port variants, each port supporting a single PRI trunk connection. Through the IP Office configuration the ports can be switched between supporting E1, E1R2 or T1 line types as required for the trunk being connected. Each port supports 8 B-channels ("Bearer") which do not need to be licensed. If additional B-channels are required, they must be licensed through the addition of the licenses below. Multiple licenses can be added to achieve the total number of channels required. D-channels are not subject to licensing. The licenses are consumed by those channels which are configured as being 'in service', in order of the installed IP500 PRI-U cards.
  - IP500 Universal PRI Additional E1 Channels Licenses
    These licenses are used to enable additional E1 B-channels for IP500 PRI-U cards configured for E1 operation. Each port can support up to 30 B-channels in this mode.
    - IP500 Universal PRI Additional Channels - 2 E1 Channels License: *IPO LIC IP500 E1 ADD 2CH (215183)*
    - IP500 Universal PRI Additional Channels - 8 E1 Channels License: *IPO LIC IP500 E1 ADD 8CH (215184)*
    - IP500 Universal PRI Additional Channels - 22 E1 Channels License: *IPO LIC IP500 E1 ADD 22CH (215185)*
  - IP500 Universal PRI Additional E1R2 Channel Licenses
    These licenses are used to enable additional E1R2 B-channels for IP500 PRI-U cards configured for E1R2 operation. Each port can support up to 30 B-channels in this mode.
    - IP500 Universal PRI Additional Channels - 2 E1R2 Channels License: *IPO LIC IP500 E1R2 ADD 2CH (215186)*
    - IP500 Universal PRI Additional Channels - 8 E1R2 Channels License: *IPO LIC IP500 E1R2 ADD 8CH (215187)*
    - IP500 Universal PRI Additional Channels - 22 E1R2 Channels License: *IPO LIC IP500 E1R2 ADD 22CH (215188)*
  - IP500 Universal PRI Additional T1 Channel Licenses
    These licenses are used to enable additional T1 B-channels for IP500 PRI-U cards configured for T1 PRI or T1 robbed-bit operation. Each port can support up to 23 B-channels (T1 PRI) or 24 B-channels (T1 robbed bit) in these mode.
    - IP500 Universal PRI Additional Channels - 2 T1 Channels License: *IPO LIC IP500 T1 ADD 2CH (215180)*
    - IP500 Universal PRI Additional Channels - 8 T1 Channels License: *IPO LIC IP500 T1 ADD 8CH (215181)*
    - IP500 Universal PRI Additional Channels - 32 T1 Channels License: *IPO LIC IP500 T1 ADD 22CH (215182)*
- IP500 Voice Networking Licenses
  The use of private voice networking trunks between IP500 control units and other systems requires voice networking channel licenses within the IP500. This applies to H323 IP trunks configured on the IP500 including IP trunks being used for an IP Office Small Community Network (SCN). A voice networking license is also required for the use of trunks configured to for QSIG operation. On H323 IP trunk, a license instance is consumed for each simultaneous outgoing call (incoming calls do NOT consume a license). On QSIG trunk, the number of calls is limited by the trunk type rather than available licenses.
  - IP500 Voice Networking (Additional Channels) License: *IPO LIC IP500 VCE NTWKG ADD 4 LIC:CU (205650)*
    This license enables 4 voice networking channels including the Advanced Networking features (distributed groups and hot-desking across Small Community Network). Additional licenses can be added to achieve the number of voice networking channels required.

SIP Trunk Channel Licenses

These licenses are used to enable channels on SIP trunks added to the IP Office system. When a channel is used it requires a license. Multiple licenses can be combined for the total number of channels required. Note that SIP calls will require VCM channels during call setup and may require VCM channels during the call.

- SIP Trunk Channels - 1 Channel: *IPO LIC SIP TRNK RFA 1 (202967)*
- SIP Trunk Channels - 5 Channels: *IPO LIC SIP TRNK RFA 5 (202968)*
- SIP Trunk Channels - 10 Channels: *IPO LIC SIP TRNK RFA 10 (202969)*
- SIP Trunk Channels - 20 Channels: *IPO LIC SIP TRNK RFA 20 (202970)*

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IP Endpoint Licenses

With Release 7.0 all IP500 IP500 V2 IP endpoints (Avaya IP endpoints as well as 3rd party IP endpoints) require an endpoint license. There are two different types of licenses available

- Avaya IP Endpoint License – required for all Avaya endpoints such as 1600, 5600, 9600 and DECT 3720/25.
- 3rd Party IP Endpoint License – this is a renaming of the IP End Point License and is required for all 3rd party SIP and H.323 endpoints.
- For each IP500 VCM32 and IP500 VCM64 card installed in the system, the 4 unlicensed VCM channels previously provided by each card are converted to allow unlicensed support of 12 Avaya IP telephones.

- Avaya IP Endpoint Licenses
  All Avaya IP500/IP500v2 IP telephones (including DECT and WiFi telephones) require an Avaya IP Endpoint license and will consume one instance of that license. Note this does not apply to the Avaya IP Office Softphone.
  - Avaya IP Endpoint License - 1 Phone: *IPO LIC R6 AV IP ENDPOINT 1 LIC:CU (229444)*
  - Avaya IP Endpoint License - 5 telephones: *IPO LIC R6 AV IP ENDPOINT 5 LIC:CU (229445)*
  - Avaya IP Endpoint License - 20 telephones: *IPO LIC R6 AV IP ENDPOINT 20 LIC:CU (229447)*
- 3rd Party IP Endpoint Licenses
  Non-Avaya IP telephones (SIP or H.323 devices) require a 3rd Party IP Endpoint license to register to IP Office and will consume one instance of that license. Note that Avaya cannot guarantee operation beyond making and receiving calls.
  - IP Endpoint License - 1 Phone: *IPO LIC IP40 IP ENDPOINT RFA 1 LIC:CU (174956)*
  - IP Endpoint License - 5 telephones: *IPO LIC IP ENDPOINT RFA 5 LIC:CU (174957)*
  - IP Endpoint License - 20 telephones: *IPO LIC IP ENDPOINT RFA 20 LIC:CU (174959)*

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Other System Licenses

- IPSec Tunneling License: IPO LIC IPSEC VPN RFA LIC:DS (182301)
  IP Office control units can be configured to support VPN tunnels using IPSec and or L2TP. While L2TP tunnels can be configured without requiring a license, the use of IPSec tunnels requires this license. Note that IP Office VPN tunnels are not suitable terminating VPN links for Avaya VPNremote client telephones.
- Centralized Intuity Audix Voicemail License: IPO LIC ACM CENTRAL VM LIC:DS (177467)
  Enables the use of a centralized Intuity Audix or Modular Messaging system for the IP Office voicemail server.
- CTI TAPI Licenses
  The IP Office TAPI software can be run in two modes; 1st party or 3rd party. 1st Party mode does not require a license and allows individual users to control their phone using a TAPI compliant application on their PC. 3rd Party mode requires licenses and allows a TAPI application to control calls for any users on the IP Office.
  - CTI Link Pro License: *IPO LIC CTI RFA LIC:DS (171988)*
    This license allows the IP Office TAPI software to be run in 3rd party mode. A license is required for each server configured to run the IP Office TAPI software in 3rd party mode.
  - WAVE User License: *IPO LIC TAPI WAV RFA 4 LIC:CU (177466)*
    Applications accessing the IP Office TAPI software in 3rd party mode can also be used to stream WAV files to users. This license allows streaming to up to 4 users at any time. Multiple licenses can be combined for a higher number of user up to the maximum supported by the IP Office control unit; IP406 V2 = 20, IP412 = 30, IP500 = 40.

Trial Licenses

The following trial licenses can be requested. Each is valid for 60 days from the date of issue and can only be issued once for a particular IP Office Feature Key serial number. Apart from that restriction the trial license works the same as a full license.

- Preferred Edition: *IPO LIC PREFERRED (VM PRO) TRIAL RFA LIC:DS (189782).*
- Advanced Edition: *IPO LIC R6 ADV EDITION TRIAL LIC:DS (229425) Release 6 onwards.*
- Power User (Users): *IPO LIC R6 PWR USER 5 TRIAL LIC:CU (229429) Release 6 onwards.*
- Customer Service Agent: *IPO LIC CUSTMR SVC AGT RFA TRIAL 5 (227053).*
- Customer Service Supervisor: *IPO LIC R6 CUSTMR SVC SPV 1 TRIAL (229443) Release 6 onwards.*
- Teleworker (Users): *IPO LIC R6 TELEWORKER 5 TRIAL:CU (229433) Release 6 onwards.*
- Mobile Worker (Users): *IPO LIC R6 MOBILE WORKER 5 TRIAL CU (229437) Release 6 onwards.*
- Office Worker (Users): *IPO LIC R6 OFF WORKER 5 TRIAL CU (229441) Release 6 onwards.*
- Receptionist (Users): *IPO LIC RECEPTIONIST RFA 1 TRIAL LIC:CU (189783).*
- VMPro Networked Messaging: *IPO LIC NTWKD MSGING TRIAL RFA LIC:DS (189776).*
- VMPro TTS (ScanSoft): *IPO LIC AVAYA TTS TRIAL RFA 1 LIC:CU (189778).*
- Audix Voicemail: *IPO LIC ACM CENTRAL VM TRIAL LIC:DS (189786).*
- IPSec Tunneling: *IPO LIC IPSec VPN RFA TRIAL LIC:DS (189806).*
- SIP Trunk Channels: *IPO LIC SIP TRUNKING TRIAL RFA 5 (205820).*
- IP500 Voice Networking: *IPO LIC IP500 VCE NTWK ADD 4 TRIAL (205823).*
- Avaya IP End Points: *IPO LIC R6 AV IP ENDPOINT 5 TRIAL (229449) Release 6 onwards.*
- Third-Party IP End Points: *IPO LIC IP ENDPOINT RFA 5 TRIAL (227040).*
- CTI Link Pro License Trial: *IPO LIC CTI RFA TRIAL (263128).*

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Licensing Examples

Some features require a combination of licenses.

Example 1: IP500 Small Community Network

	Each IP500
Required Licenses	• IP500 Voice Networking (4 Channels).
Optional Licenses	• IP500 Voice Networking (Additional Channels).

Example 2: Messaging in an IP500 Small Community Network

	Central IP500	Remote IP500
Required Licenses	• IP500 Voice Networking (4 Channels). • Preferred Edition (Messaging 4 Ports).	• IP500 Voice Networking (4 Channels).
Optional Licenses	• IP500 Voice Networking (Additional Channels). • Additional Messaging (ports).	• IP500 Voice Networking (Additional Channels). • Preferred Edition (Messaging 4 Ports). • Additional Messaging (ports).

Example 3: ContactStore Recording Library in an IP500 Small Community Network

	Central IP500	Remote IP500
Required Licenses	• IP500 Voice Networking (4 Channels). • Preferred Edition (Messaging 4 Ports). • Advanced Edition or Messaging Recordings Administrator.	• IP500 Voice Networking (4 Channels). • Advanced Edition or Messaging Recordings Administrator.
Optional Licenses	• IP500 Voice Networking (Additional Channels). • Additional Messaging (ports).	• IP500 Voice Networking (Additional Channels). • Preferred Edition (Messaging 4 Ports). • Additional Messaging (ports).

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Chapter 22. Appendix

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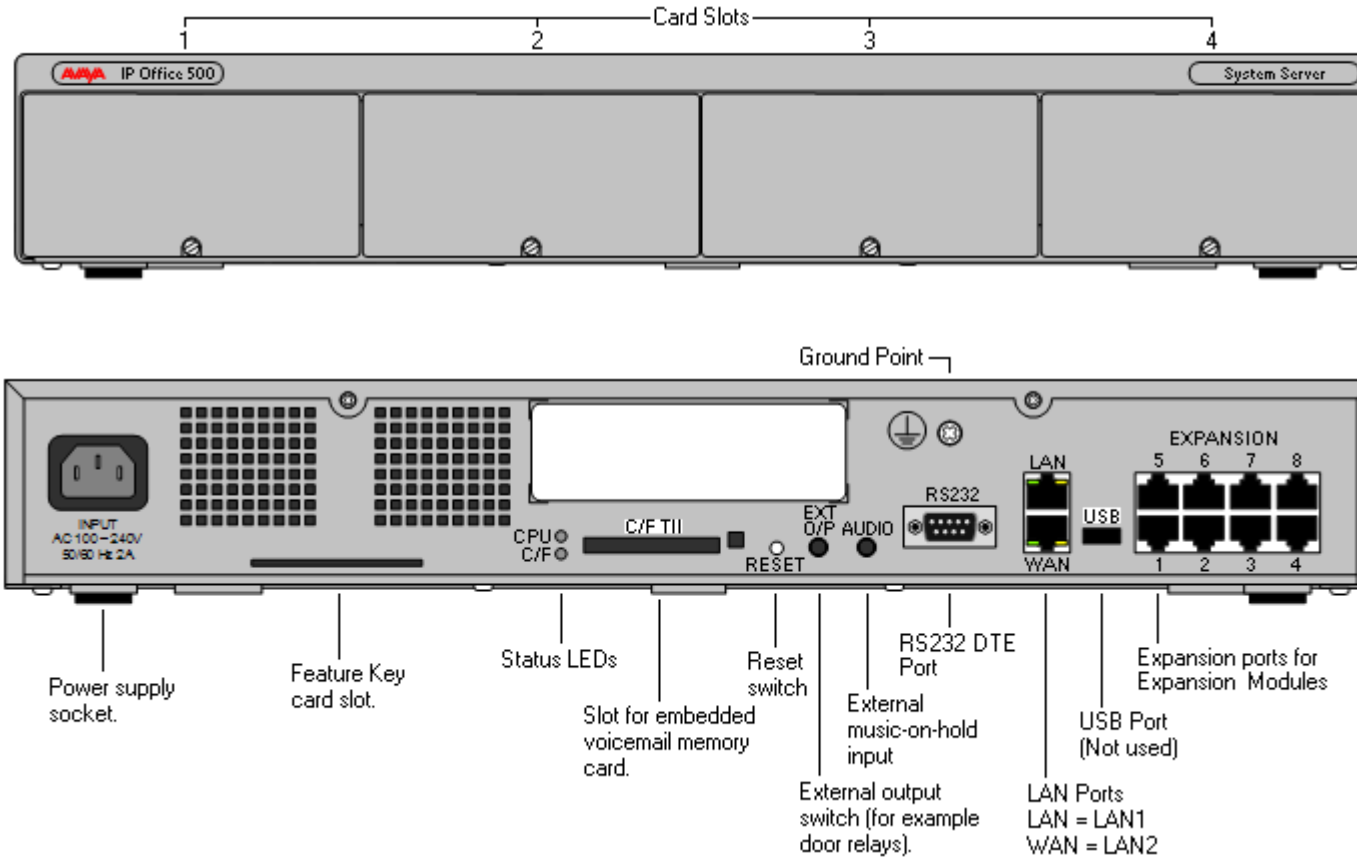
22. Appendix

The IP 500 control unit is no longer sold by Avaya. IP 500 V2 control unit is current model available.

IP 500 control units is supported with IP Office Release 6.0, 6.1 and IP Office Release 7.0 software.



The following equipment is no longer sold by Avaya but is still supported with IP Office Release 6.0 and 6.1 software

IP412 Control Unit

With a greater internal data transfer capability than the IP406 V2, the IP412 is more suitable for meeting the needs of the small contact center or businesses with a CRM focus. The IP412 differs from the IP4060 V2 by providing a greater trunk expansion capability of up to four PRI trunks. The IP412 is a stackable unit with an optional 19" rack mounting kit. The IP412 includes:

- 9-pin DTE Port (for maintenance or Feature Key connection for application licensing).
- X.21/V35 WAN interface.
- Support for up to 12 IP Office Expansion Modules:
  - Phone modules (8, 16, 30)
  - Digital Station modules (16, 30)
  - Analog Trunk Module 16
  - So8 module
- External O/P socket supporting two relay on/off switch ports, e.g. for door entry systems.
- Audio input port for external music on hold source.
- Two trunk interface card slots for analog, BRI, PRI (T1, E1) or CAS (E1R2)
- 2 internal sockets for IP Telephony expansion – voice compression modules (from 4 to 30 channels)
- Internal socket for internal modem (2 or 12) for Remote Access Services
- 108 Data channels
- Up to 30 VoiceMail Pro ports

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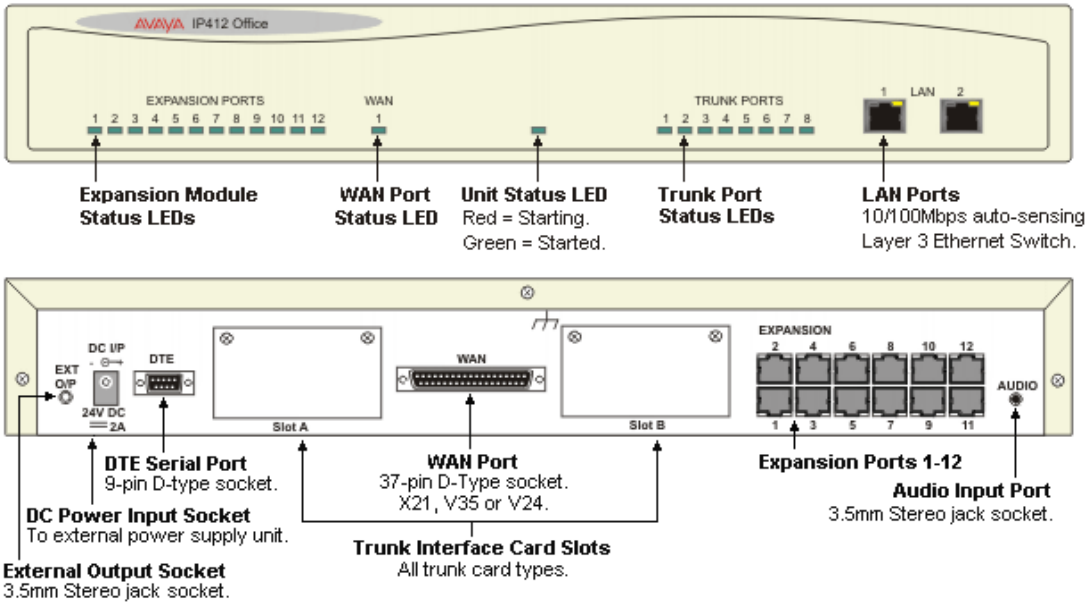
- Two 10/100 switched Ethernet ports (Layer 3).



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Expansion Modules

Through support of up to twelve external Expansion Modules.

If additional analog trunks are required, these can be aggregated in groups of 16 on each analog expansion module. It can be enhanced to support a mixture of analog, digital or IP telephones, to maximum of 3840 telephones in any combination.

Data Channels

A Data Channel is used for Remote Access (RAS), Internet Access, and Voicemail sessions. A data channel is an internal signaling resource used whenever a call is made from the IP network to an exchange line (Central Office). For example, four people surfing the Internet will use a single data channel since they all share the same line to the ISP. Two people remotely accessing the Office LAN from home will use two data channels since they have dialed in on separate lines. IP extensions do not use data channels. Data channels are used for voicemail connections with a maximum of 40 available for VoiceMail Pro.

Modems and Voice Compression modules

You can add additional hardware to the IP412 system to add one modem card (12 V.90 modems) and 2 Voice Compression Modules (VCM). Each VCM supports from 4 to 30 simultaneous Voice over IP sessions and is used for either providing networking between sites over a Wide Area Network or supporting IP Telephones and Soft phones.

IP406 V2 Control Unit

The IP406 V2 control unit is a stackable unit with an optional 19" rack mounting kit. The IP406 V2 includes:

- Eight Digital Station (DS) ports for supported 2400, 4400, 5400, 6400 and T3 Series phones plus 3810 wireless (US) phones.
- Two Analog telephone ports.
  - Two Wire
  - DTMF signaling (No rotary or Loop Disconnect)
  - Timed Break Recall (No Earth Recall)
  - Caller ID capable – a variety of standards, see later
  - MWI capable – 82.5V and Line Reversal
- Eight 10/100 Mbps LAN Switched ports (Layer-2, unmanaged).
- Support for optional embedded voicemail/auto-attendant (Compact Flash card)
- 9-pin DTE Port (for maintenance or Feature Key connection for application licensing).
- X.21/V35 WAN interface.
- Support for up to 6 IP Office Expansion Modules:
  - Phone modules (8, 16, 30)
  - Digital Station modules (16, 30)
  - Analog Trunk Module 16
  - So8 module
- External O/P socket supporting two relay on/off switch ports, e.g. for door entry systems.
- Audio input port for external music on hold source.
- Two trunk interface card slots for analog, BRI, PRI (T1, E1) or CAS (E1R2)
- Internal socket for IP Telephony expansion – voice compression modules (from 4 to 30 channels)
- Internal socket for internal modem (2 or 12) for Remote Access Services
- 50 Data channels
- Up to 20 VoiceMail Pro ports

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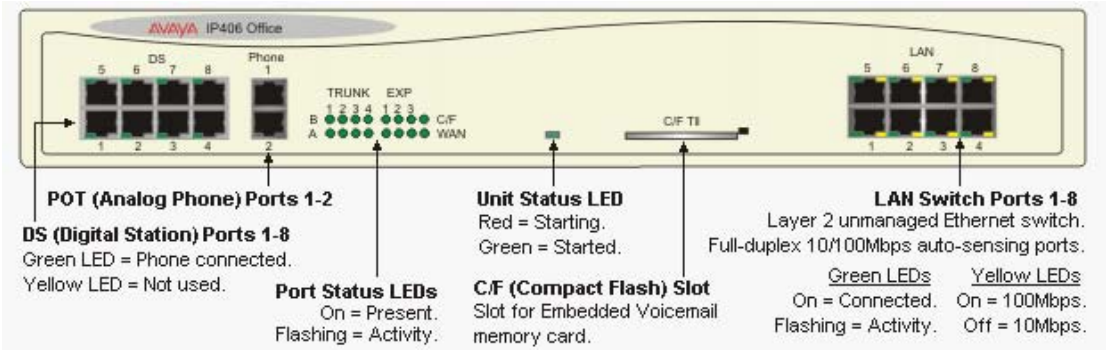

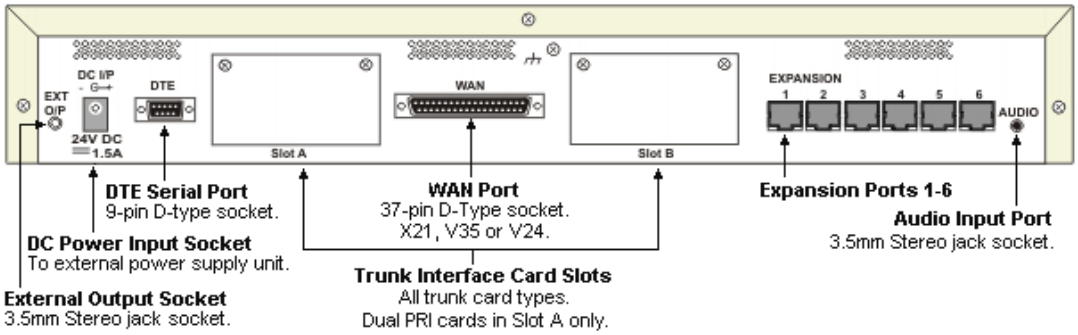

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Expansion Modules

Through support of up to six external Expansion Modules, IP406 can be enhanced to support a mixture of analog, digital or IP phones, to maximum of 190 phones in any combination.

If additional analog trunks are required, these can be aggregated in groups of 16 on each analog expansion module

Data Channels

A Data Channel is used for Remote Access (RAS), Internet Access, and Voicemail sessions. A data channel is an internal signaling resource used whenever a call is made from the IP network to an exchange line (Central Office). For example, four people surfing the Internet will use a single data channel since they all share the same line to the ISP. Two people remotely accessing the Office LAN from home will use two data channels since they have dialed in on separate lines. IP extensions do not use data channels. Data channels are used for voicemail connections with a maximum of 20 available for VoiceMail Pro on an IP406 V2.

Modems and Voice Compression modules

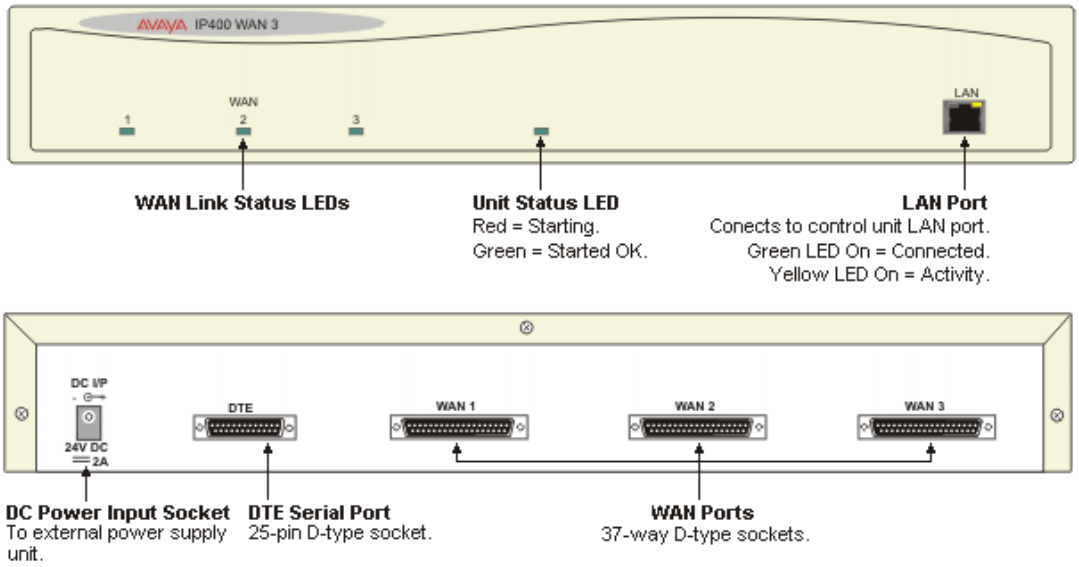
You can add additional hardware to the IP406 system to add one modem card (2 or 12 V.90 modems) and 1 Voice Compression Module (VCM). The VCM supports from 4 to 30 simultaneous Voice over IP sessions and is used for either providing networking between sites over a Wide Area Network or supporting IP Telephones and Soft phones.

IP400 WAN3 10/100

All IP400 control units except IP500 and IP500 V2

The IP400 WAN3 10/100 module provides three WAN connections (X21, V35 or V24 via a 37way D Type socket and using an appropriate connector cable). Data rates of up to 2 Mbps are supported on each interface, the carrier providing the line dictates the actual operating speed i.e. in some territories the maximum speed may be limited to 1.544 Mbps. These WAN interfaces are identical to the single WAN connection provided as standard on the IP406 and IP412 platforms.

The IP400 WAN3 10/100 connects to the control unit through the Local Area Network via a 10/100Mbps connection and does not use an expansion port on the control unit. IP500 does not support the WAN3 10/100. All other platforms support up to two WAN3 10/100 modules.



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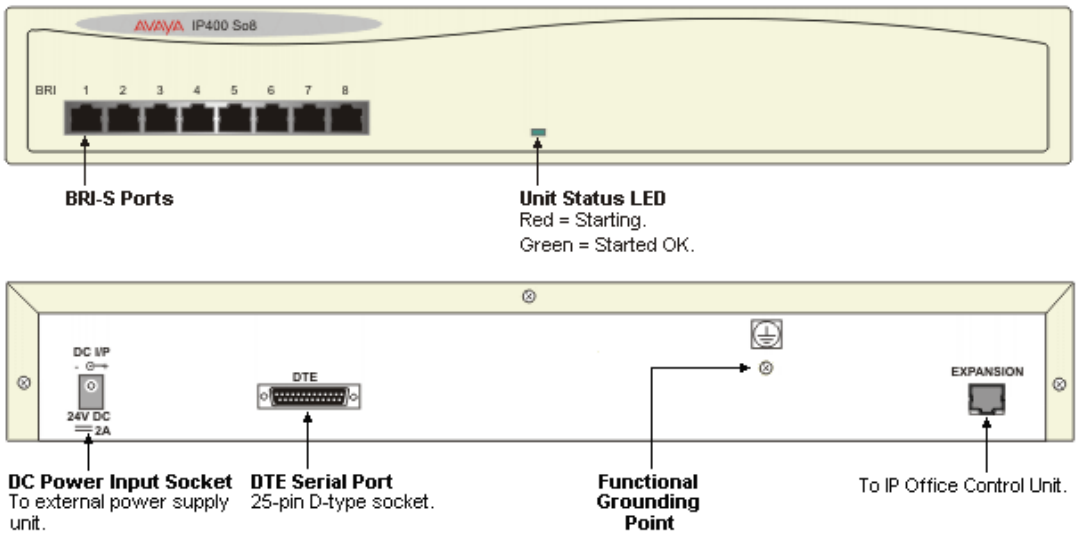
IP400 So8 Expansion Module

The IP400 So8 module provides 8 S-Bus interfaces for Basic Rate ISDN devices, such as video conferencing, fax servers or ISDN telephones.

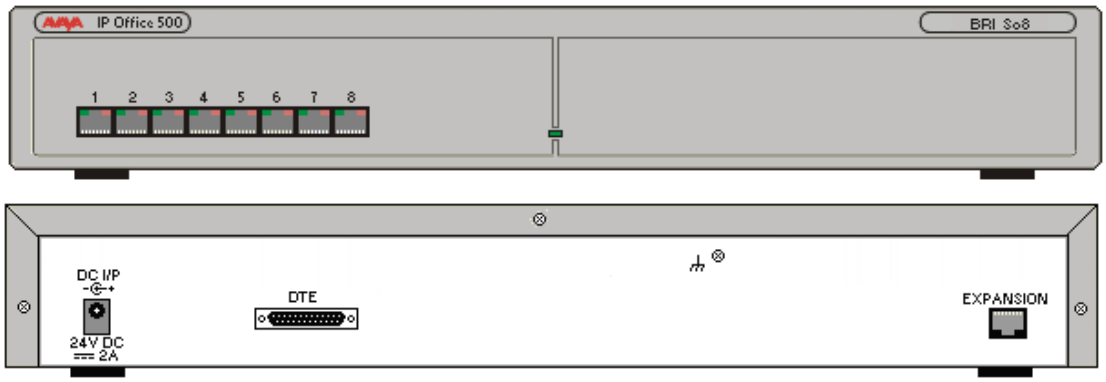


The IP Office So8 expansion module supports both point-to-point and point-to-multipoint connections. A maximum of 10 terminal endpoints identifiers (TEIs) are supported on each bus.

IP500 BRI So8 Module

The IP500 BRI So8 module provides 8 S-Bus interfaces for Basic Rate ISDN devices, such as video conferencing, fax servers or ISDN telephones.

For installations in a rack, this module requires the IP500 Rack Mounting Kit. The IP500 BRI So8 Module is functionally identical to the IP400 So8 Module.



The IP500 BRI So8 expansion module supports both point-to-point and point-to-multipoint connections. A maximum of 10 terminal endpoints identifiers (TEIs) are supported on each bus.

These modules can be used to add additional ports to an IP400 and IP500 systems. The number of external expansion modules supported depends on the control unit type. Each module uses an external power supply unit supplied with the module. A locale specific power cord for the PSU must be ordered separately.

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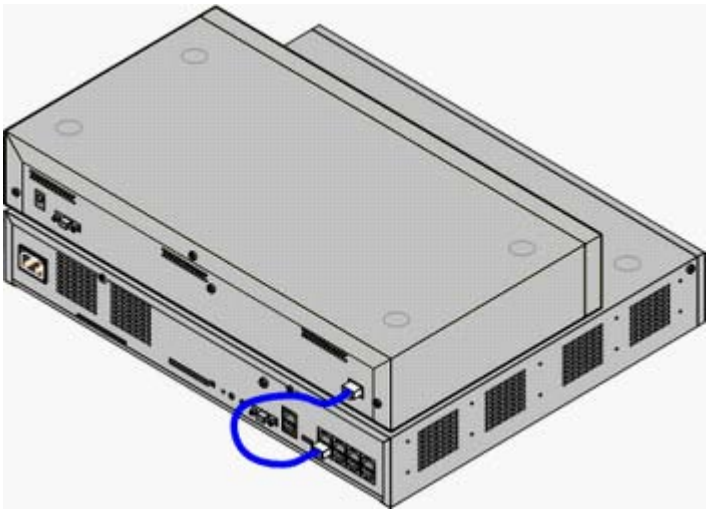
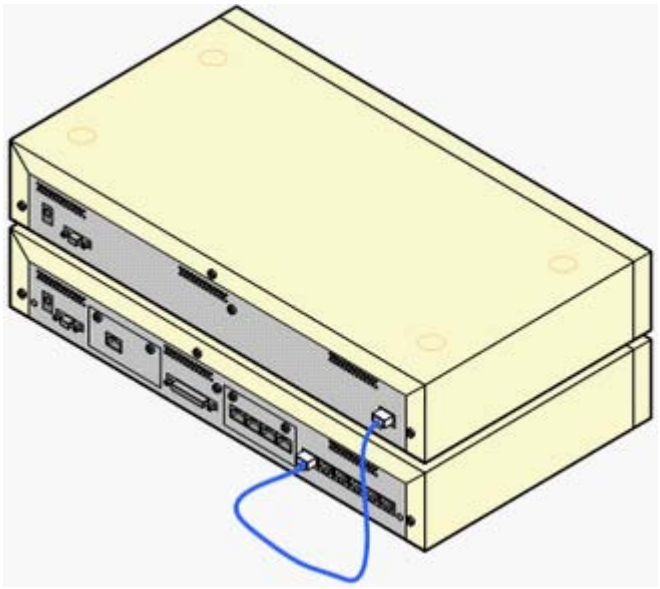


IP500 System with External Expansion Module



IP400 System with External Expansion Module

IP406 V2	6	✓
IP412	12	✓
IP500	8	–
" +4-Port Expansion Card	12	–
IP500v2	8	–
" +4-Port Expansion Card	12	–

- Each external expansion module (except the WAN3 10/100) is supplied with a blue 1 meter (3'3'') expansion interconnect cable. This cable must be used when connecting to expansion ports on the rear of a control unit.

When connecting to expansion ports on an IP500 4-Port Expansion card, a yellow 2 meter (6'6") expansion interconnect cable can be used in place of the standard blue cable. 4 Yellow cables are supplied with the

- IP500 4-Port Expansion card.
The WAN3 10/100 module differs in that it connects via a LAN port on its front to one of the control unit's
- LAN ports. This type of external expansion module is not supported by IP500 and IP500v2 control units.

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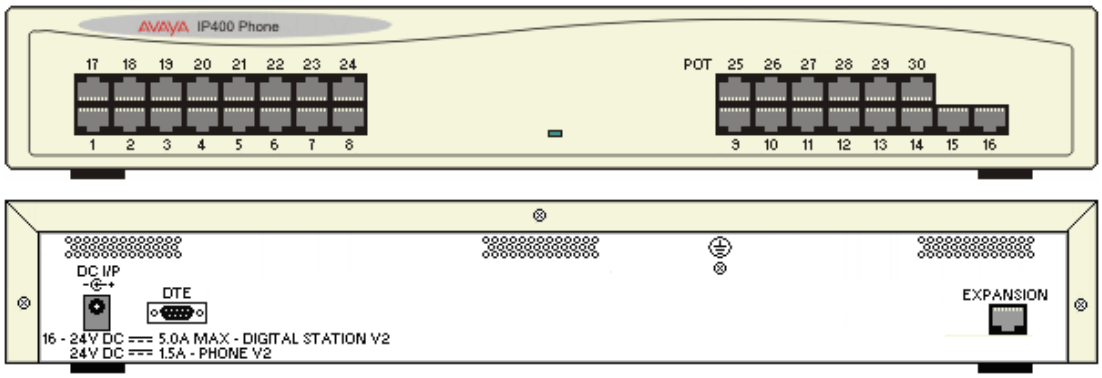
IP400 Phone Module

This module provides additional analog telephone interfaces:

- Two Wire
- DTMF signaling (No rotary or Loop Disconnect)
- Time Break Recall (No Earth Recall)
- Caller ID capable
- Message Waiting Indication (MWI) capable - High Voltage, Pulsed High Voltage, Line Reversal

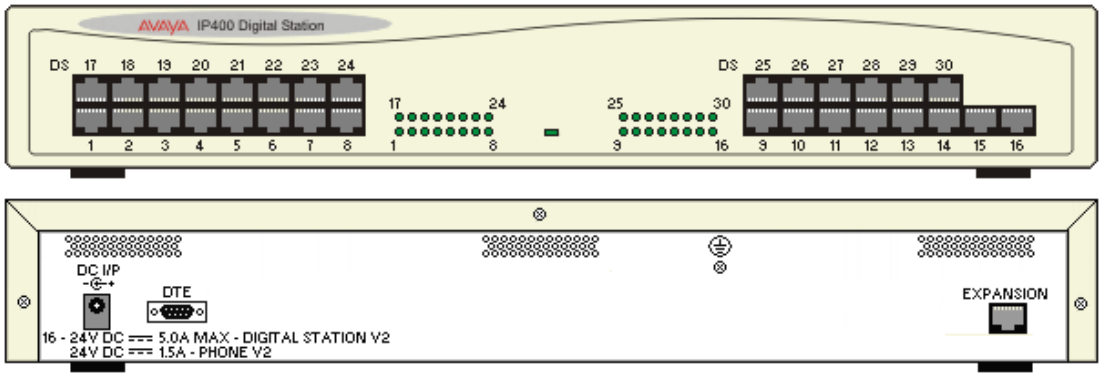
The IP400 Phone module is available in 3 versions, giving 8, 16 or 30 extensions. Telephones can be located up to 1km from the control unit. For extensions located "out-of-the-building" additional line protection will be needed. For more information on cabling and out of building guidelines, see the IP Office Installation Manual.



- IP Office Phone Modules provide support for a variety of analog MWI methods. These methods are 51V Stepped, 81V, 101V and Line Reversal. The 101V method is only supported when using a Phone V2 expansion module.
- Each analog port can support a device of maximum 1 REN.
- On analog ports, call information is sent while the phone is ringing, and cannot be updated during a call or set on an outbound call (the phone may do a local match but this is not controlled by the IP Office). The primary purpose of displays is to give information about incoming calls. Where the Caller Display standard chosen supports the delivery of text (extension name) as well as the number, both are delivered.
- An analog extension port can be set for external Paging operation. It does not operate like a normal extension and is connected to external equipment through an isolation device. The Port will always be busy so it cannot be called directly and can only be accessed by using a shortcode. When not receiving a Page the port will remain silent, when being paged the page tone is sent before the speech path is opened.

IP400 Digital Station V2 Module

This expansion module provides additional Digital Station (DS) ports for selected Avaya 2400, 4400, 5400, 6400, T3 (EMEA only) series phones and 3810 wireless phones (NA only). The IP400 Digital Station module is available in 2 variants; 16 or 30 extensions.



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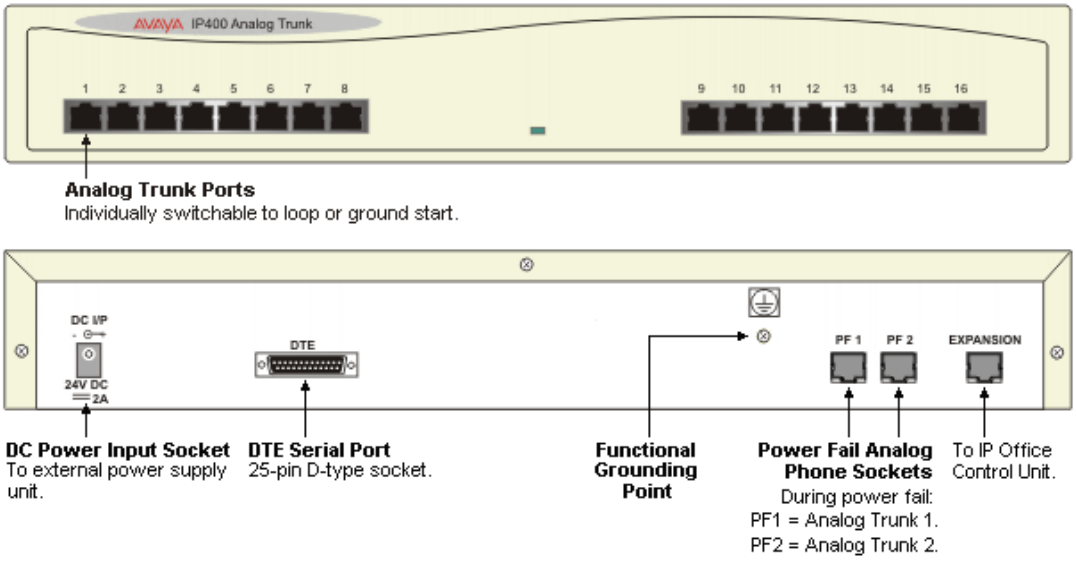
- Telephones can be located up to 1km from the control unit. For extensions located "out-of-the-building" additional line protection will be needed. For more information on cabling and out of building guidelines, see the IP Office Installation Manual.
- For systems where Direct Station Select (DSS) Units are being used, IP Office supports a maximum of:
  - Eight EU24 and or EU24BL per system.
  - Two XM24 units on each Digital Station expansion module to a maximum of 10 x XM24 units per system.
  - Two 4450 units on each Digital Station expansion module to a maximum of 8 x 4450 units per system.
  - T3 DSS units.

See the Telephones Section for specific limits on the number of each type of telephone supported on DS modules.

IP400 Analog Trunk 16 Module

This expansion module provides an additional sixteen Loop Start or Ground Start two-wire analog trunks. (Ground start trunks are not available in all territories) The first two trunks on the module which are automatically switched to power fail sockets on the rear of the unit in the event of power being interrupted must be loop start for correct power fail operation.



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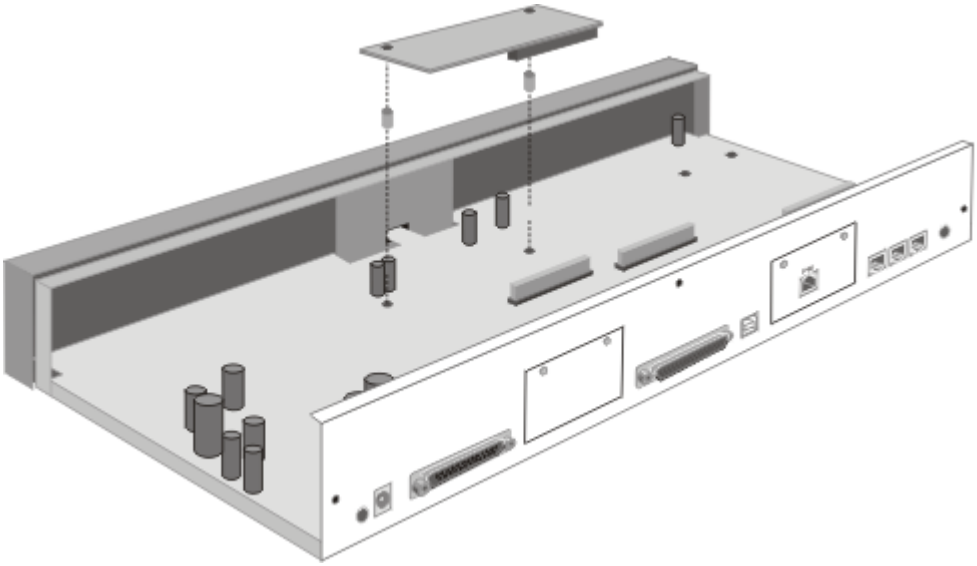
Internal Daughter Cards

Internal Daughter Cards are fitted inside the IP406 V2 and IP412 control units.



IP400 Voice Compression Module – 4/8/16/24/30 ports

The Voice Compression Module (VCM) is used for Voice over IP (VoIP) applications in the IP406 and IP412 control units. Five VCM variants are available supporting 4, 8, 16, 24 and 30 channels of compression. The echo cancellation capabilities of the VCM cards vary. The VCM 4, 8, 16 and 24 cards offer 64ms of echo cancellation while the VCM 30 card offers 25ms.

The IP406 V2 supports a single VCM while the IP412 can have any two VCMs installed.

The following table shows how many of each cards are supported by each platform.

VCM Cards*		SAP Code	IP406 V2	IP412	IP500	IP500v2
25ms echo cancellation.	IP400 VCM5	700185119	✓	✓	–	–
	IP400 VCM10	700185127	✓	✓	–	–
	IP400 VCM20	700185135	✓	✓	–	–
	IP400 VCM30	700293939	✓	✓	✓	✓
64ms echo cancellation.	IP400 VCM4	700359854	✓	✓	✓	✓
	IP400 VCM8	700359862	✓	✓	✓	✓
	IP400 VCM16	700359870	✓	✓	✓	✓
	IP400 VCM24	700359888	✓	✓	✓	✓
Number of IP400 VCM cards.			1	2	2	2
Maximum number of channels.			30	60	128	148

*Each card requires the use of an IP500 Legacy Card Carrier.

IP400 Internal Modem Card

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An internal modem card with 12 modems can be installed in both the IP406 V2 and IP412 to provide dial-up capacity that is better matched to remote access requirements of customers. The Internal Modem card allows up to 12 simultaneous V.90 (IP500 External Expansion Modules 56kbps) analog modem calls into the IP Office.

IP400 External Expansion Modules

Expansion modules include an external power supply unit (PSU) and a 1m blue interconnect cable. They do not include a locale specific power cord for the external PSU or any phone extension cables.

Variant	Country	SAP Code	IP406 V2	IP412	IP500	IP500 +4-Port Exp	IP500v2	IP500v2 +4-Port Exp
Analog Trunk								
IP400 Analog Trunk 16	North America	700211360	✓	✓	✓	–	✓	–
	Europe	700241680	✓	✓	✓	–	✓	–
	New Zealand	700241698	✓	✓	✓	–	✓	–
IP400 Digital Station 16 V2	All	700359839	✓	✓	✓	✓	✓	✓
IP400 Digital Station 30 V2	All	700359847	✓	✓	✓	✓	✓	✓
IP400 Digital Station 16	All	700184807	✓	✓	✓	–	✓	–
IP400 Digital Station 30	All	700184880	✓	✓	✓	–	✓	–
IP400 Phone 8 V2	All	700359896	✓	✓	✓	✓	✓	✓
IP400 Phone 16 V2	All	700359904	✓	✓	✓	✓	✓	✓
IP400 Phone 30 V2	All	700359912	✓	✓	✓	✓	✓	✓
IP400 Phone 8	All	700184773	✓	✓	✓	–	✓	–
IP400 Phone 16	All	700184781	✓	✓	✓	–	✓	–
IP400 Phone 30	All	700184799	✓	✓	✓	–	✓	–
IP400 So8	All	700185077	✓	✓	✓	✓	✓	✓
IP400 WAN3	All	700185028	✓	✓	–	–	–	–
IP400 WAN3 10/100	All	700262009	✓	✓	–	–	–	–

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